

PÂTENT COOPERATION TREATY

WO 99/65017
PCT/JP99/03064

PCT

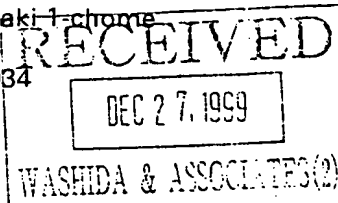
NOTICE INFORMING THE APPLICANT OF THE COMMUNICATION OF THE INTERNATIONAL APPLICATION TO THE DESIGNATED OFFICES

(PCT Rule 47.1(c), first sentence)

From the INTERNATIONAL BUREAU

To:

WASHIDA, Kimihito
Shintoshicenter Building
5th floor
24-1, Tsurumaki 1-chome
Tama-shi
Tokyo 206-0034
JAPON



Date of mailing (day/month/year) 16 December 1999 (16.12.99)		
Applicant's or agent's file reference 1F99035-PCT		
IMPORTANT NOTICE		
International application No. PCT/JP99/03064	International filing date (day/month/year) 08 June 1999 (08.06.99)	Priority date (day/month/year) 09 June 1998 (09.06.98)
Applicant MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD. et al		

1. Notice is hereby given that the International Bureau has communicated, as provided in Article 20, the international application to the following designated Offices on the date indicated above as the date of mailing of this Notice:
 CN,EP,JP,KR,US

In accordance with Rule 47.1(c), third sentence, those Offices will accept the present Notice as conclusive evidence that the communication of the international application has duly taken place on the date of mailing indicated above and no copy of the international application is required to be furnished by the applicant to the designated Office(s).

2. The following designated Offices have waived the requirement for such a communication at this time:
 CA

The communication will be made to those Offices only upon their request. Furthermore, those Offices do not require the applicant to furnish a copy of the international application (Rule 49.1(a-bis)).

3. Enclosed with this Notice is a copy of the international application as published by the International Bureau on 16 December 1999 (16.12.99) under No. WO 99/65017

REMINDER REGARDING CHAPTER II (Article 31(2)(a) and Rule 54.2)

If the applicant wishes to postpone entry into the national phase until 30 months (or later in some Offices) from the priority date, a **demand for international preliminary examination** must be filed with the competent International Preliminary Examining Authority before the expiration of 19 months from the priority date.

It is the applicant's sole responsibility to monitor the 19-month time limit.

Note that only an applicant who is a national or resident of a PCT Contracting State which is bound by Chapter II has the right to file a demand for international preliminary examination.

REMINDER REGARDING ENTRY INTO THE NATIONAL PHASE (Article 22 or 39(1))

If the applicant wishes to proceed with the international application in the **national phase**, he must, within 20 months or 30 months, or later in some Offices, perform the acts referred to therein before each designated or elected Office.

For further important information on the time limits and acts to be performed for entering the national phase, see the Annex to Form PCT/IB/301 (Notification of Receipt of Record Copy) and Volume II of the PCT Applicant's Guide.

The International Bureau of WIPO 34, chemin des Colombettes 1211 Geneva 20, Switzerland Facsimile No. (41-22) 740.14.35	Authorized officer <p style="text-align: center;">J. Zahra</p> Telephone No. (41-22) 338.83.38
--	---

PCT

INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference 1F99035-PCT	FOR FURTHER ACTION see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.	
International application No. PCT/JP 99/03064	International filing date (day/month/year) 08/06/1999	(Earliest) Priority Date (day/month/year) 09/06/1998
Applicant MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD. et al.		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

☒ It is also accompanied by a copy of each prior art document cited in this report.

1. Basis of the report

- a. With regard to the **language**, the international search was carried out on the basis of the international application in the language in which it was filed, unless otherwise indicated under this item.

☐ the international search was carried out on the basis of a translation of the international application furnished to this Authority (Rule 23.1(b)).

- b. With regard to any **nucleotide and/or amino acid sequence** disclosed in the international application, the international search was carried out on the basis of the sequence listing:

☐ contained in the international application in written form.

☐ filed together with the international application in computer readable form.

☐ furnished subsequently to this Authority in written form.

☐ furnished subsequently to this Authority in computer readable form.

☐ the statement that the subsequently furnished written sequence listing does not go beyond the disclosure in the international application as filed has been furnished.

☐ the statement that the information recorded in computer readable form is identical to the written sequence listing has been furnished

2. ☐ **Certain claims were found unsearchable** (See Box I).

3. ☐ **Unity of invention is lacking** (see Box II).

4. With regard to the **title**,

☒ the text is approved as submitted by the applicant.

☐ the text has been established by this Authority to read as follows:

5. With regard to the **abstract**,

☒ the text is approved as submitted by the applicant.

☐ the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this international search report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is Figure No.

☐ as suggested by the applicant.

☒ because the applicant failed to suggest a figure.

☐ because this figure better characterizes the invention.

5

☐ None of the figures.

INTERNATIONAL SEARCH REPORT

International Application No

JP 99/03064

A. CLASSIFICATION OF SUBJECT MATTER

IPC 6 G10L9/14

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 95 16260 A (PACIFIC COMM SCIENCES INC) 15 June 1995 (1995-06-15) abstract; claims 1-7	1, 5, 13, 15
A	---	17
Y	WO 91 13432 A (UNIV SHERBROOKE) 5 September 1991 (1991-09-05) abstract page 2, line 20 - line 31 page 14, line 1 - line 21 ---	1, 5, 13, 15
	-/--	

☒ Further documents are listed in the continuation of box C.☒ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

"E" earlier document but published on or after the international filing date

"L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)

"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search

21 September 1999

Date of mailing of the international search report

28/09/1999

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2
 NL - 2280 HV Rijswijk
 Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
 Fax: (+31-70) 340-3016

Authorized officer

Van Doremalen, J

INTERNATIONAL SEARCH REPORT

International Application No

JP 99/03064

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	AKAMINE M ET AL: "ADAPTIVE DENSITY PULSE EXCITATION FOR LOW BIT RATE SPEECH CODING" IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS, COMMUNICATIONS AND COMPUTER SCIENCES, vol. E78-A, no. 2, 1 February 1995 (1995-02-01), pages 199-207, XP000515192 ISSN: 0916-8508 abstract page 199, right-hand column, line 43 - line 49	1,6,11, 13,15,17
A	--- MASAMI AKAMINE ET AL: "CELP CODING WITH AN ADAPTIVE DENSITY PULSE EXCITATION MODEL" SPEECH PROCESSING 1, ALBUQUERQUE, APRIL 3 - 6, 1990, vol. 1, no. CONF. 15, 3 April 1990 (1990-04-03), pages 29-32, XP000146401 INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS abstract paragraph '02.2! -----	1,6,11, 13,15,17

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/JP 99/03064

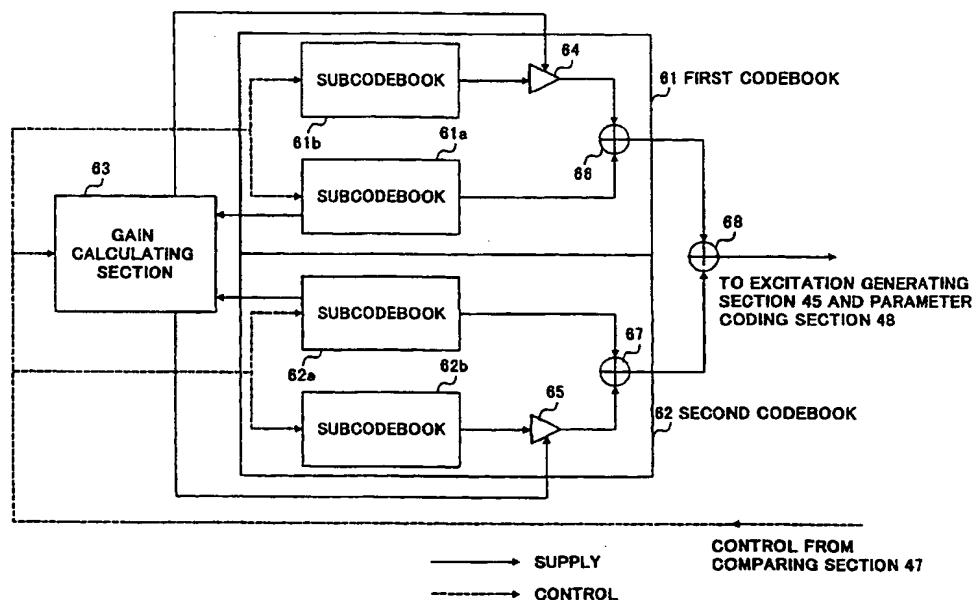
Patent document cited in search report	Publication date	Patent family member(s)	Publication date
W0 9516260 A	15-06-1995	US 5717824 A	10-02-1998
		AU 1336995 A	27-06-1995
		CA 2178073 A	15-06-1995
		EP 0733257 A	25-09-1996
W0 9113432 A	05-09-1991	CA 2010830 A	23-08-1991
		AT 164252 T	15-04-1998
		AU 6632890 A	18-09-1991
		DE 69032168 D	23-04-1998
		DE 69032168 T	08-10-1998
		EP 0516621 A	09-12-1992
		ES 2116270 T	16-07-1998
		US 5699482 A	16-12-1997
		US 5754976 A	19-05-1998
		US 5701392 A	23-12-1997
		US 5444816 A	22-08-1995



INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(51) International Patent Classification ⁶ : G10L 9/14		A1	(11) International Publication Number: WO 99/65017
			(43) International Publication Date: 16 December 1999 (16.12.99)
(21) International Application Number: PCT/JP99/03064		(81) Designated States: CA, CN, JP, KR, US, European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).	
(22) International Filing Date: 8 June 1999 (08.06.99)			
(30) Priority Data: 10/160119 9 June 1998 (09.06.98) JP 10/258271 11 September 1998 (11.09.98) JP		Published <i>With international search report.</i>	
(71) Applicant (for all designated States except US): MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD. [JP/JP]; 1006, Oaza Kadoma, Kadoma-shi, Osaka 571-8501 (JP).			
(72) Inventors; and			
(75) Inventors/Applicants (for US only): MORII, Toshiyuki [JP/JP]; 2-3-7-501, Nijigaoka, Asao-ku, Kawasaki-shi, Kanagawa 215-0015 (JP). YASUNAGA, Kazutoshi [JP/JP]; 3-33-17-305, Sugao, Miyamae-ku, Kawasaki-shi, Kanagawa 216-0015 (JP).			
(74) Agent: WASHIDA, Kimihito; Shintoshicenter Building, 5th floor, 24-1, Tsurumaki 1-chome, Tama-shi, Tokyo 206-0034 (JP).			

(54) Title: SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS



(57) Abstract

First codebook (61) and second codebook (62) respectively have two subcodebooks, and in respective codebooks, addition sections (66) and (67) obtain respective excitation vectors by adding sub-excitation vectors fetched from respective two subcodebooks. Addition section (68) obtains an excitation sample by adding those excitation vectors. According to the aforementioned constitution, it is possible to store sub-excitation vectors with different characteristics in respective subcodebooks. Therefore, it is possible to correspond to input signals with various characteristics, and achieve excellent sound qualities at the time of decoding.

FOR THE PURPOSES OF INFORMATION ONLY

Codes used to identify States party to the PCT on the front pages of pamphlets publishing international applications under the PCT.

AL	Albania	ES	Spain	LS	Lesotho	SI	Slovenia
AM	Armenia	FI	Finland	LT	Lithuania	SK	Slovakia
AT	Austria	FR	France	LU	Luxembourg	SN	Senegal
AU	Australia	GA	Gabon	LV	Latvia	SZ	Swaziland
AZ	Azerbaijan	GB	United Kingdom	MC	Monaco	TD	Chad
BA	Bosnia and Herzegovina	GE	Georgia	MD	Republic of Moldova	TG	Togo
BB	Barbados	GH	Ghana	MG	Madagascar	TJ	Tajikistan
BE	Belgium	GN	Guinea	MK	The former Yugoslav Republic of Macedonia	TM	Turkmenistan
BF	Burkina Faso	GR	Greece	ML	Mali	TR	Turkey
BG	Bulgaria	HU	Hungary	MN	Mongolia	TT	Trinidad and Tobago
BJ	Benin	IE	Ireland	MR	Mauritania	UA	Ukraine
BR	Brazil	IL	Israel	MW	Malawi	UG	Uganda
BY	Belarus	IS	Iceland	MX	Mexico	US	United States of America
CA	Canada	IT	Italy	NE	Niger	UZ	Uzbekistan
CF	Central African Republic	JP	Japan	NL	Netherlands	VN	Viet Nam
CG	Congo	KE	Kenya	NO	Norway	YU	Yugoslavia
CH	Switzerland	KG	Kyrgyzstan	NZ	New Zealand	ZW	Zimbabwe
CI	Côte d'Ivoire	KP	Democratic People's Republic of Korea	PL	Poland		
CM	Cameroon	KR	Republic of Korea	PT	Portugal		
CN	China	KZ	Kazakhstan	RO	Romania		
CU	Cuba	LC	Saint Lucia	RU	Russian Federation		
CZ	Czech Republic	LI	Liechtenstein	SD	Sudan		
DE	Germany	LK	Sri Lanka	SE	Sweden		
DK	Denmark	LR	Liberia	SG	Singapore		
EE	Estonia						

DESCRIPTION

SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS

Technical Field

5 The present invention relates to a speech coding apparatus and a speech decoding apparatus using speech coding algorithm at low bit rates, used in digital communications such as a portable telephone.

10 Background Art

 Speech compression coding methods at low bit rates have been required in order to accept an increase of subscribers in digital mobile communications such as a portable telephone, and the researches and developments have been proceeded by many
15 research institutions. In Japan, applied coding systems as a standard system in portable telephones are VSELP at a bit rate of 11.2kbps developed by Motorola and PSI-CELP at a bit rate of 5.6kbps developed by NTT Mobile Communications Network, INC., and portable telephones with these system are produced.

20 In addition, internationally, the ITU-T selected CS-ACELP, which was co-developed by Nippon Telegraph and Telephone Corporation and France Telecom, as an international standard speech coding system G.729 at 8kbps. The system is scheduled to be used in Japan as speech coding system for
25 portable telephones.

 The above-described systems are all achieved by modifying the CELP system (Code Excited Linear Prediction:

M.R.Schroeder "High Quality Speech at Low Bit Rates" described in Proc.ICASSP '85 pp.937-940). A feature of this system is to apply a method of dividing a speech into excitation information and vocal tract information, code the excitation information with indices of a plurality of excitation samples stored in a codebook, while coding the LPC (Linear Prediction Coefficients) with respect to the vocal tract information, and perform a comparison to an input speech considering of the vocal tract information in the excitation information coding (A-
10 b-S: Analysis by Synthesis).

The basic algorithm of the CELP system will be described using FIG.1. FIG.1 is a block diagram illustrating a configuration of a speech coding apparatus in the CELP system. In the speech coding apparatus illustrated in FIG.1, LPC
15 analyzing section 2 executes autocorrelation analysis and LPC analysis on input speech data 1 to obtain the LPC. LPC analyzing section 2 further codes the obtained LPC to obtain the coded LPC. LPC analyzing section 2 furthermore decodes the obtained coded LPC to obtain the decoded LPC.

20 Excitation generating section 5 fetches excitation samples stored in adaptive codebook 3 and stochastic codebook 4 (respectively referred to as an adaptive code vector (or adaptive excitation) and stochastic code vector (or stochastic excitation)) and provides respective excitation samples to LPC
25 synthesis section 6. LPC synthesis section 6 executes filtering on two excitations obtained at excitation generating section 5 with the decoded LPC obtained at LPC analyzing section

2.

Comparing section 7 analyzes the relation of two synthesized speeches obtained at LPC synthesis section 6 and the input speech, obtains an optimal value (optimal gain) for two synthesized speeches, adds each synthesized speech respectively subjected to power adjustment with the optimal gain to obtain a total synthesized speech, and executes a distance calculation between the total synthesized speech and the input speech. Comparing section 7 further executes, with respect to all excitation samples in adaptive codebook 3 and stochastic codebook 4, the distance calculations between the input speech and each of other many synthesized speeches obtained by functioning excitation generating section 5 and LPC synthesis section 6, and obtains an index of the excitation sample whose distance is the smallest among the obtained distances. Then, comparing section 7 provides the obtained optimal gain, indices of excitation samples of respective codebooks and two excitation samples corresponding to respective index to parameter coding section 8.

Parameter coding section 8 executes coding on the optimal gain to obtain the coded gain and provides the coded gain, the coded LPC and the indices of excitation samples to transmission path 9. Further, parameter coding section 8 generates an actual excitation signal (synthesized excitation) using the coded gain and two excitations corresponding to the respective index and stores the excitation signal in adaptive codebook 3 while deleting old excitation samples.

In addition, it is general for the synthesis at LPC synthesis section 6 to use together Linear Prediction Coefficients and a high-frequency enhancement filter or a perceptual weighting filter with long-term prediction coefficients (which are obtained by the long-term prediction analysis of input speech). It is further general to execute the excitation search on the adaptive codebook and stochastic codebook at an interval (called subframe) obtained by further dividing an analysis interval.

10 The stochastic codebook will be described next.

 The adaptive codebook is a codebook for an effective compression using a long-term correlation existing at intervals of human vocal cord vibrations, and stores previous synthesized excitations. On the contrary, the stochastic codebook is a fixed codebook to reflect statistical characteristics of excitation signals. As excitation samples stored in the stochastic codebook, there are, for example, random number sequence, pulse sequence, random number sequence/pulse sequence obtained by statistic training with speech data, or pulse sequence with relatively small number of pulses generated algebraically (algebraic codebook). The algebraic codebook has been especially paid attention recently and known by that a good sound quality is obtained at bit rates such as 8kbps with small calculation amounts .

25 However, an application of algebraic codebook with a small number of pulses to coding at lower bit rates introduces a phenomenon that sound qualities greatly deteriorate mainly

on unvoiced consonants and background noises. On the other hand, an application of excitation with a large number of pulses such as random number sequence to coding at lower bit rates introduces a phenomenon that sound qualities greatly deteriorate mainly on voiced speeches. In order to improve the deterioration, a method with multi-codebook, in which a voiced/unvoiced judgement is performed, is examined. However, the method has the complicated processing and sometimes generates an allophone caused by a judgement error on a speech signal.

As described above, there has been no algebraic codebook which matches any effective coding on voiced speeches, unvoiced speeches and background noises. Therefore, it has been required to obtain a speech coding apparatus and a speech decoding apparatus capable of effectively coding any of voiced speeches, unvoiced speeches and background noises.

Disclosure of Invention

An object of the present invention is to provide a speech coding apparatus and a speech decoding apparatus capable of effectively coding any of voiced speeches, unvoiced speeches and background noises and obtaining speeches with excellent qualities with a small amount of information and a small amount of computations.

The inventors of the present invention noticed that pulse positions are relatively near at a voiced sound segment of speech, while pulse positions are relatively far at segments

of unvoiced sound of speech and background noise, in the case of applying a pulse sequence to coding at low bit rates. In other words, the inventors noticed that energy-concentrated excitation samples, which are characteristics of human vocal cord wave, are needed in a voiced speech, and in this case, there is a tendency that a small number of pulses whose positions are near are selected, while an excitation having more random number characteristics is needed in a unvoiced speech and background noise, in this case, there is a tendency that a large number of energy-spread pulses are selected.

Based on the foregoing consideration, the inventors found out that the perception is improved by identifying a speech as voiced sound segment, or unvoiced sound segment and background noise segment by recognizing a distance of pulse positions, and based on the identification result, applying respective pulse sequences appropriate for the voiced sound segment, and the unvoiced and background noise segments, to achieve the present invention.

A feature of the present invention is to use a plurality of codebooks each having two subcodebook of which characteristics are different and add excitation vectors of each subcodebook to obtain excitation vectors. According to the algorithm, the characteristics as a small-number-pulse excitation appear in the case where pulse positions are near, which is caused by the relationships of positions of the excitation vectors with a small number of pulses, while the characteristics as a large-number-pulse excitation appear in

the case where pulse positions are far, which is suited to the characteristics of speech signals containing background noises.

Accordingly, without using particular voiced/unvoiced
5 speech judgement algorithm, it is possible to automatically select an excitation most suitable for the localized characteristics in input signals, effectively code any of voiced speeches, unvoiced speeches and background noises, and obtain synthesized speeches with excellent sound qualities
10 with a small amount of information and a small amount of computations.

Brief Description of Drawings

FIG.1 is a block diagram illustrating a configuration
15 of a speech coding apparatus in a conventional CELP system;

FIG.2 is a block diagram illustrating a configuration of a radio communication apparatus having a speech coding apparatus and a speech decoding apparatus of the present invention;

20 FIG.3 is a block diagram illustrating a configuration of a speech coding apparatus in a CELP system according to a first embodiment to a third embodiment of the present invention;

FIG.4 is a block diagram illustrating a configuration
25 of a speech decoding apparatus in the CELP system according to a first embodiment to a third embodiment of the present invention;

FIG.5 is a block diagram illustrating a stochastic codebook in a speech coding apparatus/speech decoding apparatus according to the first embodiment of the present invention;

5 FIG.6A and FIG.6B are concept diagrams of sub-excitation vectors stored in subcodebooks in the stochastic codebook;

FIG.s 7A to 7F are concept diagrams to explain a generation method of excitation sample;

10 FIG.8 is a block diagram illustrating a stochastic codebook in a speech coding apparatus/speech decoding apparatus according to the second embodiment of the present invention;

15 FIG.9 is a block diagram illustrating a stochastic codebook in a speech coding apparatus/speech decoding apparatus according to the third embodiment of the present invention;

FIG.10A and FIG.10B are concept diagrams of sub-excitation vectors stored in subcodebooks in the stochastic codebook;

20 FIG.s 11A to 11F are concept diagrams to explain a generation method of excitation sample; and

FIG.12 is a diagram illustrating a schematic configuration of a data medium storing a program for the speech coding apparatus/speech decoding apparatus of the present invention.

25

Best Mode for Carrying Out the Invention

Embodiments of the present invention will be described in detail with reference to accompanying drawings.

(First embodiment)

FIG.2 is a block diagram illustrating a configuration of a radio communication apparatus having a speech coding/decoding apparatus according to the first embodiment to the third embodiment of the present invention.

In this radio communication apparatus, at a transmitting side, a speech is converted into electric analogue signals at speech input device 21 such as a microphone and output to A/D converter 22. The analogue speech signals are converted into digital speech signals at A/D converter 22 and output to speech coding section 23. Speech coding section 23 executes speech coding processing on the digital speech signals and outputs the coded data to modulation/demodulation circuit 24. Modulation/demodulation circuit 24 executes digital modulation on the coded speech signals to output to radio transmission circuit 25. Radio transmission circuit 25 executes the predetermined radio transmission processing on the modulated signals. The signals are transmitted via antenna 26. In addition, processor 31 executes the processing properly using data stored in RAM 25 and ROM 26.

On the other hand, at a receiving side in the radio communication apparatus, received signals received at antenna 26 are subjected to the predetermined radio reception processing at radio reception circuit 27 and output to modulation/demodulation circuit 24.

Modulation/demodulation circuit 24 executes demodulation processing on the received signals and outputs the demodulated signals to speech decoding section 28. Speech decoding section 28 executes decoding processing on the demodulated signals to obtain digital decoded speech signals and output the digital decoded speech signals to D/A converter 29. D/A converter 29 converts the digital decoded speech signals output from speech decoding section 28 into analogue decoded speech signals to output to speech output device 30 such as a speaker. Finally, speech output device 30 converts electric analogue decoded speech signals into decoded speech to output.

Speech coding section 23 and speech decoding section 28 are operated by processor 31 such as DSP using codebooks stored in RAM 32 and ROM 33. The operation program is also stored in ROM 33.

FIG.3 is a block diagram illustrating a configuration of a speech coding apparatus in the CELP system according to the first embodiment to the third embodiment of the present invention. The speech coding apparatus is included in speech coding section 23 illustrated in FIG.2. In addition, adaptive codebook 43 illustrated in FIG.3 is stored in RAM 32 illustrated in FIG.2, and stochastic codebook 44 illustrated in FIG.3 is stored in ROM 33 illustrated in FIG.2.

In the speech coding apparatus (hereinafter, also referred to as coder) illustrated in FIG.3, LPC analyzing section 42 executes autocorrelation analysis and LPC analysis on input speech data 41 to obtain the LPC. LPC analyzing

section 42 further codes the obtained LPC to obtain the LPC code. LPC analyzing section 42 furthermore decodes the obtained LPC code to obtain the decoded LPC. In the coding, it is generally executed to convert into parameters having good
5 interoperation characteristics such as LSP (Linear Spectrum Pair) then code by VQ (Vector Quantization).

Excitation generating section 45 fetches excitation samples stored in adaptive codebook 43 and stochastic codebook 44 (respectively referred to as adaptive code vector (or
10 adaptive excitation) and stochastic code vector (or stochastic excitation)) and provides respective excitation samples to LPC synthesis section 46. The adaptive codebook is a codebook in which excitation signals previously synthesized are stored and an index represents which synthesized excitation is used among
15 from excitations synthesized at different previous times, i.e., time lag.

LPC synthesis section 46 executes filtering on two excitations obtained at excitation generating section 45 with the decoded LPC obtained at LPC analyzing section 42.

20 Comparing section 47 analyzes the relation of two synthesized speeches obtained at LPC synthesis section 46 and the input speech, obtains an optimal value (optimal gain) for two synthesized speeches, adds each synthesized speech respectively subjected to power adjustment with the optimal
25 gain to obtain a total synthesized speech, and executes a distance calculation between the total synthesized speech and the input speech. Comparing section 47 further executes, with

respect to all excitation samples in adaptive codebook 43 and stochastic codebook 44, the distance calculations between the input speech and each of other many synthesized speeches obtained by functioning excitation generating section 45 and
5 LPC analyzing section 46, and obtains an index of the excitation sample whose distance is the smallest among the obtained distances. Then, comparing section 47 provides the obtained optimal gain, indices of excitation samples of respective codebooks and two excitation samples corresponding to
10 respective index to parameter coding section 48.

Parameter coding section 48 executes coding on the optimal gain to obtain the gain code and provides the gain code, the LPC code and the indices of excitation samples to transmission path 49. Further, parameter coding section 48
15 generates an actual excitation signal (synthesized excitation) using the gain code and two excitations corresponding to the index and stores the excitation signal in adaptive codebook 43 while deleting old excitation samples.

In addition, it is general for the synthesis at LPC
20 synthesis section 46 to use together Linear Prediction Coefficients and a high-frequency enhancement filter or a perceptual weighting filter with long-term prediction coefficients (which are obtained by the long-term prediction analysis of input speech). It is further general to execute
25 the excitation search on the adaptive codebook and stochastic codebook at an interval (called subframe) obtained by further dividing an analysis interval.

FIG.4 is a block diagram illustrating a configuration of a speech decoding apparatus in the CELP system according to the first embodiment to the third embodiment of the present invention. The speech decoding apparatus is included in
5 speech decoding section 28 illustrated in FIG.2. In addition, adaptive codebook 53 illustrated in FIG.4 is stored in RAM 32 illustrated in FIG.2, and stochastic codebook 54 illustrated in FIG.4 is stored in ROM 33 illustrated in FIG.2.

In the speech decoding apparatus illustrated in FIG.4,
10 parameter decoding section 52 obtains coded speech signals from transmission path 51, while obtains respective coded excitation samples of excitation codebooks (adaptive codebook 53 and stochastic codebook 54), the coded LPC and coded gain. Parameter decoding section 52 then obtains the decoded LPC
15 using the coded LPC and the decoded gain using the coded gain.

Excitation generating section 55 multiplies each excitation sample respectively by the decoded gain to obtain decoded excitation signals. At this stage, excitation generating section 55 stores the obtained decoded excitation
20 signals in adaptive codebook 53 as excitation samples, while deletes old excitation samples. LPC synthesis section 56 executes filtering on the decoded excitation signals with the decoded LPC to obtain a synthesized speech.

In addition, these two excitation codebooks are the same
25 as those included in the speech coding apparatus illustrated in FIG.3 (reference numerals 43 and 44 in FIG.3). Sample numbers to fetch excitation samples (codes to adaptive

codebook and codes to stochastic codebook) are both supplied from parameter decoding section 52 (which corresponds to the short dashes line in FIG.5 (control from comparing section 47) described later).

5 The following description is given to explain in detail about functions of stochastic codebooks 44 and 54 for storing excitation samples, in the speech coding apparatus and speech decoding apparatus with the above configurations, using FIG.5
10 FIG.5 is a block diagram illustrating a stochastic codebook in the speech coding apparatus and speech decoding apparatus according to the first embodiment of the present invention.

 The stochastic codebook has first codebook 61 and second codebook 62, and first codebook 61 and second codebook 62 respectively have two subcodebooks 61a, 61b and 62a, 62b. The
15 stochastic codebook further has gain calculating section 63 which calculates a gain for outputs from subcodebooks 61b and 62b using pulse positions in subcodebooks 61a and 62a.

 Subcodebooks 61a and 62a are mainly used in the case where a speech is a voiced sound (pulse positions are relatively near),
20 and formed by storing a plurality of sub-excitation vectors composed of a single pulse. Subcodebook 61b and 62b are mainly used in the case where a speech is an unvoiced sound or background noise (pulse positions are relatively far), and formed by storing a plurality of sub-excitation vectors
25 composed of a sequence with a plurality of pulses in which power is spread. The excitation samples are generated in the stochastic codebooks formed as described above. In addition,

the near and far pulse positions will be described later.

In addition, subcodebooks 61a and 62b are formed by a method of arranging pulses algebraically, and subcodebooks 61b and 62b are formed by another method of dividing a vector length
5 (subframe length) into some segment intervals and making a configuration so that a single pulse is always present at every segment interval (pulses are spread over a whole length).

These codebooks are formed in advance. In this embodiment, as illustrated in FIG.5, the number of codebooks
10 is set at two and each codebook has two subcodebooks.

FIG.6A illustrates sub-excitation vectors stored in subcodebook 61a of first codebook 61. FIG.6B illustrates sub-excitation vectors stored in subcodebook 61b of first codebook 61. Similarly, subcodebooks 62a and 62b of second
15 codebook 62 respectively have sub-excitation vectors illustrated in FIG.6A and FIG.6B.

In addition, positions and polarities of pluses of sub-excitation vectors in subcodebooks 61b and 62b are formed using random numbers. According to the configuration
20 described above, it is possible to form sub-excitation vectors in which power is uniformly spread over a whole vector length even though some fluctuations are present. FIG.6B illustrates an example in the case where the number of segment intervals is four. In addition, in these two subcodebooks, respective
25 sub-excitation vectors of the same index (number) are used at the same time.

The next description is given to explain speech coding

using the stochastic codebooks with the above-mentioned configuration.

Gain calculating section 63 calculates an excitation vector number (index) according to the code from comparing section 47 in the speech coding apparatus. The code provided from comparing section 47 corresponds to the excitation vector number, and therefore the excitation vector number is determined by the code. Gain calculating section 63 fetches sub-excitation vectors with a small number of pulses corresponding to the determined excitation vector number from subcodebooks 61a and 62a. Gain calculating section 63 further calculates an addition gain using pulse positions of the fetched sub-excitation vectors. The addition gain calculation is given by the following equation (1);

$$g = |P_1 - P_2| / L \quad \dots \text{equation (1)}$$

where g is an addition gain, P_1 and P_2 are respectively pulse positions in codebooks 61a and 62a, and L is a vector length (subframe length). Further, $| |$ represents an absolute value.

According to the above equation (1), the addition gain is smaller as the pulse positions are nearer (the pulse distance is shorter), while larger as pulse positions are further, and has a lower limit of 0 and an upper limit of 1. Accordingly, as the pulse positions are nearer, the gain for subcodebooks 61b and 62b is relatively smaller. As a result, an affect of subcodebooks 61a and 62b corresponding to voiced speech is larger. On the other hand, as the pulse positions are further (the pulse distance is longer), the gain for subcodebooks 61b

and 62b is relatively larger. As a result, an affect of subcodebooks 61b and 62b corresponding to unvoiced speech and background noise is relatively larger. Perceptually fine sounds are obtained by performing the gain control described
5 above.

Next, gain calculating section 63 refers to the number of excitation vector provided from comparing section 47 and obtains two sub-excitation vectors from subcodebooks 61b and 62b with a large number of pulses. These two sub-excitation
10 vectors from subcodebooks 61b and 62b are respectively provided to gain calculating sections 64 and 65 to be multiplied by the addition gain obtained at gain calculating section 63.

Further, excitation vector addition section 66 obtains a sub-excitation vector from subcodebook 61a with a small
15 number of pulses by referring to the number of excitation vector provided from comparing section 47, and also obtains the sub-excitation vector, from subcodebook 61b, multiplied by the addition gain obtained at gain calculating section 63. Excitation vector addition section 66 then adds the obtained
20 sub-excitation vectors to obtain an excitation vector. Similarly, excitation vector addition section 67 obtains a sub-excitation vector from subcodebook 62a with a small number of pulses by referring to the number of excitation vector provided from comparing section 47, and also obtains the
25 sub-excitation vector, from subcodebook 62b, multiplied by the addition gain obtained at gain calculating section 63. Excitation vector addition section 67 then adds the obtained

sub-excitation vectors to obtain an excitation vector.

The excitation vectors respectively obtained by adding the sub-excitation vector are provided to excitation vector addition section 68 to be added. According to the foregoing
5 processing, an excitation sample (stochastic code vector) is obtained. The excitation sample is provided to excitation generating section 45 and parameter coding section 48.

On the other hand, a decoding side prepares the same adaptive codebook and stochastic codebook as those in the coder
10 in advance, and based on respective index, LPC code, and gain code of each codebook transmitted from the transmission path, multiplies respective excitation sample by the gain to add. Then the decoding side executes filtering on the added sample with the decoded LPC to decode the speech.

15 An example of excitation samples selected by the above-mentioned algorithm will be described next using FIG.7A to FIG.7F. Assume that an index of first codebook 61 is j , and an index of second codebook 62 is m or n .

As been understood from FIG.7A and FIG.7B, in the case
20 of $j+m$, since the pulse positions of the sub-excitation vectors of subcodebooks 61a and 62a are relatively near, a small value of the addition gain is calculated using the equation (1) described previously. Accordingly, the addition gain for subcodebooks 61b and 62b is small. Because of it, as
25 illustrated in FIG.7C, excitation vector addition section 68 obtains an excitation sample composed of a small number of pulses which reflects the characteristics of subcodebooks 61a

and 62a respectively illustrated in FIG.7A and FIG.7B. This excitation sample is effective on voiced speech.

Further, as been understood from FIG.7A and FIG.7B, in the case of $j+n$, since the pulse positions of the sub-excitation vectors of subcodebooks 61a and 62a are relatively far, a large value of the addition gain is calculated using the equation (1) described previously. Accordingly, the addition gain for subcodebooks 61b and 62b is large. Because of it, as illustrated in FIG.7F, excitation vector addition section 68 obtains an excitation sample with strong random characteristics with spread energy which reflects the characteristics of subcodebooks 61b and 62b respectively illustrated in FIG.7D and FIG.7E. This excitation sample is effective on unvoiced speech /background noise.

This embodiment describes about the case of using two codebooks (two channels). However, it is also preferable to apply the present invention to the case of using codebooks equal to or more than three (channels equal to or more than three). In this case, as a numerator of the equation in gain calculating section 63, i.e., equation (1), the minimum value among from intervals between two pulses or the averaged value of all pulse intervals is used. For example, in the case where the number of codebooks is three and the minimum pulse interval is used as a numerator of the equation (1), the calculation equation is given by the following equation (2);

$$g = \min(|P_1 - P_2|, |P_2 - P_3|, |P_3 - P_1|) / L \quad \dots \text{equation (2)}$$

where g is an addition gain, P_1 , P_2 and P_3 are respective pulse

positions in those three codebooks, and L is a vector length (subframe length). In addition, $||$ represents an absolute value.

As described above, according to this embodiment, a plurality of codebooks have two subcodebooks each having respective sub-excitation vectors of which characteristics are different, and the excitation vector is obtained by adding each sub-excitation vector, thereby making it possible to correspond to input signals with various characteristics.

In addition, since the gain to be multiplied by the sub-excitation vector is varied corresponding to the characteristics of the sub-excitation vectors, it is possible to reflect both characteristics of excitation vectors stored in two codebooks in the speech by a gain adjustment, thereby making it possible to effectively execute coding and decoding most suitable for the characteristics of the input signals with various characteristics.

Specifically, since one of two subcodebooks stores a plurality of sub-excitation vectors composed of a small number of pulses, and another subcodebook stores a plurality of sub-excitation vectors composed of a large number of pulses, it is possible to achieve fine sound qualities in voiced speech by the excitation sample with the characteristics of a small number of pulses, and perform excitation generation most suitable to the characteristics of input signals with various characteristics.

In addition, since gain calculating section calculates

a gain using a distance of pulse positions of sub-excitation vectors composed of a small number of pulses, it is possible to achieve synthesized speeches with fine sound qualities in voiced speech by the small number of pulses of which distance is near, while achieve perceptually fine synthesized speeches in unvoiced speech/background noise by the large number of pulses with spread energy.

In the addition gain calculation described above, the processing is simplified by using a fixed value which is predetermined as an addition gain. In this case, it is not necessary to install gain calculating section 63. Even in this case, it is possible to achieve synthesized speeches matching the needs timely by varying the setting of the fixed value properly. For example, it is possible to achieve coding excellent for plosive speech such as low voice like male voice by setting the addition gain on a small scale, while to achieve coding excellent for random speeches such as background noise by setting the addition gain on a large scale.

In addition, it is also preferable to apply a method of calculating an addition gain adaptively using a level of input signal power, decoded LPC or adaptive codebook, besides the method of calculating the addition gain using pulse positions and another method of providing fixed coefficients to the addition gain. For example, it may be possible to achieve excellent coding adaptive for localized speech characteristics by preparing a function for determining voiced speech characteristics (such as vowel and standing wave) or

unvoiced speech characteristics (such as background noise and unvoiced consonant) and setting a small gain in the case of voiced speech characteristics, while setting a large gain in the case of unvoiced speech characteristics.

5 (Second embodiment)

This embodiment will describes about the case where a gain calculating section obtains decoded LPC from LPC analyzing section 42 and performs a voiced/unvoiced judgement using the obtained LPC.

10 FIG.8 is a block diagram illustrating a stochastic codebook in the speech coding apparatus/speech decoding apparatus according to the second embodiment of the present invention. The configurations of the speech coding apparatus and the speech decoding apparatus with the stochastic code book
15 are the same as the first embodiment (FIG.3 and FIG.4).

The stochastic codebook has first codebook 71 and second codebook 72, and first codebook 71 and second codebook 72 respectively have two subcodebooks 71a, 71b and subcodebooks 72a, 72b. The stochastic codebook further has gain
20 calculating section 73 which calculates a gain for outputs from subcodebooks 71b and 72b using pulse positions in subcodebooks 71a and 72a.

Subcodebooks 71a and 72a are mainly used in the case where a speech is a voiced sound (pulse positions are relatively near),
25 and formed by storing a plurality of sub-excitation vectors composed of a single pulse. Subcodebook 71b and 72b are mainly used in the case where a speech is an unvoiced sound or

background noise (pulse positions are relatively far), and formed by storing a plurality of sub-excitation vectors composed of a sequence with a plurality of pulses in which power is spread. The excitation samples are generated in the
5 stochastic codebooks formed as described above.

In addition, subcodebooks 71a and 72a are formed by a method of arranging pulses algebraically, and subcodebooks 71b and 72b are formed by another method of dividing a vector length (subframe length) into some segment intervals and making a
10 configuration so that a single pulse is always present at every segment interval (pulses are spread over a whole length).

These codebooks are formed in advance. In this embodiment, as illustrated in FIG.8, the number of codebooks is set at two and each codebook has two subcodebooks. The
15 number of codebooks and the number of subcodebooks are not limited.

FIG.6A illustrates sub-excitation vectors stored in subcodebook 71a of first codebook 71. FIG.6B illustrates sub-excitation vectors stored in subcodebook 71b of first
20 codebook 71. Similarly, subcodebooks 72a and 72b of second codebook 72 respectively have sub-excitation vectors illustrated in FIG.6A and FIG.6B.

In addition, positions and polarities of pluses of sub-excitation vectors in subcodebooks 71b and 72b are formed
25 using random numbers. According to the configuration described above, it is possible to form sub-excitation vectors in which power is uniformly spread over a whole vector length

even though some fluctuations are present. FIG.6B illustrates an example in the case where the number of segment intervals is four. In addition, in these two subcodebooks, respective sub-excitation vectors of the same index (number) are used at
5 the same time.

The next description is given to explain speech coding using the stochastic codebooks with the above-mentioned configuration.

Gain calculating section 73 obtains decoded LPC from LPC
10 analyzing section 42 and performs a voiced/unvoiced judgement using the obtained LPC. Specifically, gain calculating section 73 beforehand collects data corresponding to LPC, for example, obtained by converting the LPC into impulse response or LPC cepstrum, with respect to a lot of speech data, by
15 relating to every mode, for example, voiced speech, unvoiced speech and background noise. Then the data are subjected to statistic processing and based on the result, a rule of judging voiced, unvoiced and background noise is generated. As an example of the rule, it is general to use linear determination
20 function and Bayes judgment. Then, based on the judgment result obtained according to the rule, weighting coefficient R is obtained by a regulation of the following equation (3);

$R=L$:when judged as voiced speech

$R=L \times 0.5$:when judged as unvoiced speech/

25 background noise ...equation (3)

where R is a weighting coefficient, and L is a vector length (subframe length).

Gain calculating section 73 next receives an instruction of the number of excitation vector (index number) from comparing section 47 in the speech coding apparatus, and according to the instruction, fetches sub-excitation vectors of the designated number respectively from subcodebooks 71a and 72a with a small number of pulses. Gain calculating section 73 calculates an addition gain using pulse positions of the fetched sub-excitation vectors. The calculation of the addition gain is executed according to the following equation

10 (4);

$$g = |P1 - P2| / R \quad \dots \text{equation (4)}$$

where g is an addition gain, P1 and P2 are respectively pulse positions in codebooks 71a and 72a, and R is a weighting coefficient. Further, $| \cdot |$ represents an absolute value.

15 According to the above equations (3) and (4), the addition gain is smaller as the pulse positions are nearer, while larger as pulse positions are further, and has a lower limit of 0 and an upper limit of L/R. Accordingly, as the pulse positions are nearer, the gain for subcodebooks 71b and 72b is relatively smaller. As a result, an affect of subcodebooks 71a and 72a corresponding to voiced speech is larger. On the other hand, as the pulse positions are further, the gain for subcodebooks 71b and 72b is relatively larger. As a result, an affect of subcodebooks 71b and 72b corresponding to unvoiced
20 speech and background noise is larger. Perceptually fine sounds are obtained by performing the gain calculation described above.

Further, excitation vector addition section 76 obtains a sub-excitation vector from subcodebook 61a with a small number of pulses by referring to the number of excitation vector provided from comparing section 47, and also obtains a sub-excitation vector, from subcodebook 71b, multiplied by the addition gain obtained at gain calculating section 73. Excitation vector addition section 76 then adds the obtained sub-excitation vectors to obtain an excitation vector. Similarly, excitation vector addition section 77 obtains a sub-excitation vector from subcodebook 72a with a small number of pulses by referring to the number of excitation vector provided from comparing section 47, and also obtains a sub-excitation vector, from subcodebook 72b, multiplied by the addition gain obtained at gain calculating section 73. Excitation vector addition section 77 then adds the obtained sub-excitation vectors to obtain an excitation vector.

The excitation vectors respectively obtained by adding the sub-excitation vector are provided to excitation vector addition section 68 to be added. According to the foregoing processing, an excitation sample (stochastic code vector) is obtained. The excitation sample is provided to excitation generating section 45 and parameter coding section 48.

On the other hand, a decoding side prepares the same adaptive codebook and stochastic codebook as those in the coder in advance, and based on respective index, LPC code, and gain code of each codebook transmitted from the transmission path, multiplies respective excitation sample by the gain to add.

Then the decoding side executes filtering on the added sample with the decoded LPC to decode the speech.

At this stage, it is necessary to provide the decoded LPC to the stochastic codebook in this embodiment, which
5 differs from the first embodiment. Specifically, at this stage, parameter decoding section 52 provides the obtained LPC along with the sample number for the stochastic codebook to the stochastic codebook (which corresponds to that the signal
10 line from parameter decoding section 52 to stochastic codebook 54 in FIG. 4 includes the signal line from "LPC analyzing section 42" and the control line indicative of "control from comparing section 47").

The excitation samples selected by the above algorithm are the same as the first embodiment and illustrated in FIG. 7A
15 to FIG. 7F.

As described above, according to this embodiment, gain calculating section 73 performs the voiced/unvoiced judgement using the decoded LPC, and calculates the addition gain using weighting coefficient R obtained according to
20 equation (3), resulting in a small gain at the time of voiced speech and a large gain at the time of unvoiced speech and background noise. The obtained excitation samples are thereby a smaller number of pulses in voiced speech and a large number of pulses containing more noises in unvoiced speech and
25 background noise. Accordingly, it is possible to further improve the effect by adaptive pulse positions described above, thereby enabling synthesized speech with more excellent sound

qualities to be achieved.

In addition, the speech coding in this embodiment also has the effect on transmission error. In the coding with a conventional voiced/unvoiced judgment, stochastic codebooks
5 are switched generally by LPC. Because of it, when a transmission error introduces a wrong judgment, the decoding is sometimes executed with absolutely different excitation samples, resulting in a low transmission error resistance.

On the contrary, in the speech coding in this embodiment,
10 if wrong LPC are used in the voiced/unvoiced judgment in decoding, only a value of addition gain varies a little, and the deterioration caused by the transmission error is little. Hence, according to this embodiment, it is possible to obtain synthesized speeches with excellent sound qualities without
15 being affected by the transmission error of LPC code largely, while executing the adaptation by LPC.

This embodiment describes about the case of using two codebooks (two channels). However, it is also preferable to apply the present invention to the case of using codebooks equal
20 to or more than three (channels equal to or more than three). In this case, as a numerator of the equation in gain calculating section 63, i.e., equation (4), the minimum value among from intervals between two pulses or the averaged value of all pulse intervals is used.

25 The first and second embodiments describe about the case of adjusting gains for outputs from subcodebooks 61b, 62b, 71b and 72b. However, it is also preferable to adjust outputs from

subcodebooks 61a, 62a, 71a and 72a or to adjust outputs from all subcodebooks, under the condition that a gain for outputs from subcodebooks is adjusted so that an affect by excitation vectors with a small number of pulses is large when pulse positions are near, while an affect by excitation vectors with a large number of pulses is large when pulse positions are far. (Third embodiment)

This embodiment will describe about the case of switching an excitation vector to acquire from a subcodebook corresponding to a distance of pulse intervals.

FIG.9 is a block diagram illustrating a stochastic codebook in the speech coding apparatus/speech decoding apparatus according to the third embodiment of the present invention. The configurations of the speech coding apparatus and the speech decoding apparatus with the stochastic code book are the same as the first embodiment (FIG.3 and FIG.4).

The stochastic codebook has first codebook 91 and second codebook 92, and first codebook 91 and second codebook 92 respectively have two subcodebooks 91a, 91b and subcodebooks 92a, 92b. The stochastic codebook further has excitation switching instructing section 93 which executes switching between outputs from subcodebooks 91b and 92b corresponding to a pulse position in subcodebooks 91a and 92a.

Subcodebooks 91a and 92a are mainly used in the case where a speech is a voiced sound (pulse positions are relatively near), and formed by storing a plurality of sub-excitation vectors composed of a single pulse. Subcodebook 91b and 92b are mainly

used in the case where a speech is an unvoiced sound or background noise (pulse positions are relatively far), and formed by storing a plurality of sub-excitation vectors composed of a sequence with a plurality of pulses in which power is spread. The excitation samples are generated in the stochastic codebooks formed as described above.

In addition, subcodebooks 91a and 92a are formed by a method of arranging pulses algebraically, and subcodebooks 91b and 92b are formed by another method of dividing a vector length (subframe length) into some segment intervals and making a configuration so that a single pulse is always present at every segment interval (pulses are spread over a whole length).

These codebooks are formed in advance. In this embodiment, as illustrated in FIG.9, the number of codebooks is set at two and each codebook has two subcodebooks. The number of codebooks and the number of subcodebooks are not limited.

FIG.10A illustrates sub-excitation vectors stored in subcodebook 91a of first codebook 91. FIG.10B illustrates sub-excitation vectors stored in subcodebook 91b of first codebook 91. Similarly, subcodebooks 92a and 92b of second codebook 92 respectively have sub-excitation vectors illustrated in FIG.10A and FIG.10B.

In addition, positions and polarities of pluses of sub-excitation vectors in subcodebooks 91b and 92b are formed using random numbers. According to the configuration described above, it is possible to form sub-excitation vectors

in which power is uniformly spread over a whole vector length even though some fluctuations are present. FIG.10B illustrates an example in the case where the number of segment intervals is four. In addition, in these two subcodebooks, 5 respective sub-excitation vectors of the same index (number) are not used at the same time.

The next description is given to explain speech coding using the stochastic codebooks with the above-mentioned configuration.

10 Excitation switching instructing section 93 calculates the excitation vector number (index) according to a code from comparing section 47 in the speech coding section. The code provided from comparing section 47 corresponds to the excitation vector number, and therefore the excitation vector 15 number is determined by the code. Excitation switching instructing section 93 fetches sub-excitation vectors with a small number of pulses corresponding to the determined excitation vector number from subcodebooks 91a and 92a. Further, excitation switching instructing section 93 executes 20 a judgment described as below, using pulse positions of the fetched sub-excitation vectors;

$|P_1 - P_2| < Q$:using subcodebooks 91a and 92a

$|P_1 - P_2| \geq Q$:using subcodebooks 91b and 92b,

where P_1 and P_2 are respectively pulse positions in 25 subcodebooks 91a and 92a, Q is a constant and $||$ represents an absolute value.

In the above judgment, excitation vectors with a small

number of pulses are selected when pulse positions are near, while excitation vectors with a large number of pulses are selected when pulse positions are far. Performing the judgment and selection as described above enables perceptually fine sounds to be achieved. The constant Q is predetermined. It is possible to vary the ratio of the excitation with a small number of pulses and the excitation with a large number of pulses by varying the constant Q .

Excitation switching instructing section 93 fetches excitation vectors from subcodebooks 91a and 92a or subcodebooks 91b and 92b in codebooks 91 or 92 according to the switching information (switching signal) and the code of excitation (sample number). The switching is executed at first and second switches 94 and 95.

The obtained excitation vectors are provided to excitation vector addition section 96 to be added. The excitation sample (stochastic code vector) is thus obtained. The excitation sample is provided to excitation generating section 45 and parameter coding section 48. In addition, at a decoding side, the excitation sample is provided to excitation generating section 55.

An example of excitation samples selected by the above-mentioned algorithm will be described next using FIG.11A to FIG.11F. Assume that an index of first codebook 91 is j , and an index of second codebook 92 is m or n .

As been understood from FIG.11A and FIG.11B, in the case if $j+m$, since the pulse positions of the sub-excitation vectors

of subcodebooks 91a and 92a are relatively near, excitation switching instructing section 93 selects sub-excitation vectors with a small number of pulses according to the above judgment. Then, excitation vector addition section 96 adds
5 two sub-excitation vectors selected respectively from subcodebooks 91a and 92a illustrated in FIG.11A and FIG.11B. and obtains an excitation sample with strong pulse characteristics as illustrated in FIG.11C. This excitation sample is effective on voiced speech.

10 Further, as been understood from FIG.11A and FIG.11B, in the case of $j+n$, since the pulse positions of the sub-excitation vectors of subcodebooks 91a and 92a are relatively far, excitation switching instructing section 93 selects sub-excitation vectors with a large number of pulses according
15 to the above judgment. Then, excitation vector addition section 96 adds two sub-excitation vectors selected respectively from subcodebooks 91b and 92b illustrated in FIG.11D and FIG.11E. and obtains an excitation sample with strong random characteristics with spread energy as
20 illustrated in FIG.11F. This excitation sample is effective on unvoiced speech/background noise.

As described above, according to this embodiment, an excitation sample is generated by switching excitation vectors in two subcodebooks which a plurality of codebooks each have
25 to obtain, and using excitation vectors obtained from either of subcodebooks in each codebook. It is thus possible to correspond to input signals with various characteristics by

a fewer amount of computations.

Since one of two subcodebooks stores a plurality of excitation vectors with a small number of pulses while another one stores a plurality of excitation vectors with a large number of pulses in which power is spread, it is possible to use the excitation sample with a small number of pulses for voiced speech while use another excitation sample with a large number of pluses for unvoiced speech/background noise. It is thereby possible to obtain synthesized speeches with excellent sound qualities, and also to obtain excellent performances for input signals with various properties.

Further, since the excitation switching instructing section switches excitation vectors to acquire from a subcodebook corresponding to a distance between pulse positions, it is possible to achieve synthesized speeches with fine sound qualities in voiced speech by a small number of pulses of which distances are near, wile achieve perceptually fine synthesized speeches in unvoiced speech and background noise by a large number of pulses in which power is spread. Furthermore, since the excitation switching instructing section acquires excitation vectors from a subcodebook while switching, for example, it is not necessary to calculate a gain and multiple the gain by a vector in an stochastic codebook. Accordingly, in the speech coding according to this embodiment, a computation amount is much less than the case of calculating the gain.

That is, since the above-mentioned switching is executed

based on a relative distance between pulse positions of sub-excitation vectors composed of a small number of pulses, it is possible to achieve fine synthesized speeches in voiced speech by excitation samples with a small number of pulses of which distance are near, while achieve perceptually fine synthesized speeches in unvoiced speech/background noise by excitation samples with a large number of pulses with spread power.

This embodiment describes about the case of using two codebooks (two channels). However, it is also preferable to apply the present invention to the case of using codebooks equal to or more than three (channels equal to or more than three). In this case, as a judgment basis in excitation switching instructing section 93, the minimum value among from intervals between two pulses or the averaged value of all pulse intervals is used. For example, in the case of using three codebooks and the minimum value among from intervals between two pulses, the judgment basis is as follows;

$\min(|P_1 - P_2|, |P_2 - P_3|, |P_3 - P_1|) < Q$: using subcodebooks a

$\min(|P_1 - P_2|, |P_2 - P_3|, |P_3 - P_1|) \geq Q$: using subcodebooks b

where P_1 , P_2 and P_3 are respectively pulse positions in respective codebooks, Q is a weighting coefficient, and $|$ represents an absolute value.

In the speech coding/decoding according to this embodiment, it may be possible to combine voiced/unvoiced judgment algorithm in the same way as the second embodiment. In other words, at a coding side, the excitation switching

instructing section obtains decoded LPC from the LPC analyzing section and executes the voiced/unvoiced judgment using the LPC, and at a decoding side, the decoded LPC is provided to the stochastic codebook. According to the aforementioned
5 processing, it is possible to improve the effect by adapted pulse positions and achieve synthesized speeches with more excellent sound qualities.

The above constitution is achieved by providing voiced/unvoiced judgment sections separately at a coding side
10 and a decoding side and corresponding to the judgment result, making Q variable as a threshold value for the judgment of excitation switching instructing section. In this case, Q is set at a large scale in the case of voiced speech while Q is set at a low scale in the case of unvoiced speech in order to
15 enable varying the ratio of the number of excitations with a small number of pulses and the number of excitations with a large number of pulses corresponding to localized characteristics of speeches.

In addition, in the case where the voiced/unvoiced
20 judgment is executed by backward (using other decoded parameters without transmitting as code), there is a possibility that a wrong judgment occurs by transmission error. According to the coding/decoding in this embodiment, since the voiced/unvoiced judgment is executed only by varying
25 threshold Q , a wrong judgment affects only a difference of threshold Q between in the cases of voiced speech and unvoiced speech. Accordingly, the affects caused by the wrong judgment

is very small.

In addition, it may be possible to use a level of input signal power, decoded LPC and a method of calculating Q adaptively using an adaptive codebook. For example, prepare
5 in advance a function for determining voiced characteristics (such as vowel and standing wave) or unvoiced characteristics (such as background noise and unvoiced consonant) using the above parameters, and set Q at a large scale at the time of the voiced characteristics, while set Q at a low scale at the
10 time of the unvoiced characteristics. According to the aforementioned processing, it is possible to use an excitation sample composed of a small number of pulses in a voiced characteristics interval and another excitation sample composed of a large number of pulses in a unvoiced
15 characteristics interval, thereby making it possible to obtain excellent coding performance adaptive for speech localized characteristics.

In addition, the speech coding/decoding according to the first to third embodiments are described as speech coding
20 apparatus/speech decoding apparatus, however it may be possible to construct the speech coding/decoding as software. For example, it may be possible to store the program for the above-described speech coding/decoding in a ROM and operate by instructions of a CPU according to the program. Further,
25 as illustrated in FIG.12, it may be possible to store program 101a, adaptive codebook 101b and algebraic codebook 101c in recording medium 101 which is readable by computer, write

program 101a of recording medium 101, adaptive codebook 101b and stochastic codebook 101c in a RAM of a computer and operate according to the program. These cases also achieve the same functions and effects as the first to third embodiments
5 described above.

The first to third embodiments describe the case where the number of pulses is one as an excitation vector with a small number of pulses, it may be possible to use an excitation vector in which the number of pulses is equal to or more than two as
10 an excitation vector with a small number of pulses. In this case, it is preferable to apply an interval of pulses whose positions are the nearest among from a plurality of pulses as the near-far judgment of pulse positions.

The first to third embodiments describe about the case
15 of adapting the present invention to speech coding apparatus/speech decoding apparatus in the CELP system, however the present invention is applicable to any speech coding/decoding using "codebook" because the feature of the present invention is in an stochastic codebook. For example,
20 the present invention is applicable to "RPE-LPT" that is a standard full rate codec by GSM and "MP-MLQ" that is an international standard codec "G.723.1" by ITU-T.

This application is based on the Japanese Patent Applications No.HEI10-160119 filed on June 9, 1998 and
25 No.HEI10-258271 filed on Sep. 11, 1998, entire contents of which are expressly incorporated by reference herein.

Industrial applicability

The speech coding apparatus and speech decoding apparatus according to the present invention are applicable to portable telephones and digital communications using speech
5 coding algorithm at low bit rates.

This application is based on the Japanese Patent Applications No.HEI10-160119 filed on June 9, 1998 and No.HEI10-258271 filed on September 11, 1998, entire content of which is expressly incorporated by reference herein.

CLAIMS

1. An apparatus for performing speech coding in a CELP system, said apparatus comprising:

an adaptive codebook in which previously synthesized
5 excitation signals are stored;

a stochastic codebook in which a plurality of excitation vectors are stored, said stochastic codebook having a first subcodebook in which excitation vectors composed of a small number of pulses are stored and a second subcodebook in which
10 excitation vectors composed of a large number of pulses are stored;

means for obtaining a synthesized speech using excitation information acquired from said adaptive codebook and said stochastic codebook, using LPC obtained by performing
15 LPC analysis on an input speech signal;

means for obtaining gain information for said synthesized speech using a relation of said synthesized speech and said input speech signal; and

means for transmitting said LPC, said excitation
20 information and said gain information.

2. The apparatus according to claim 1, wherein said stochastic codebook further has control means for controlling a gain for respective excitation vectors in said first subcodebook and said second subcodebook corresponding to a
25 distance between pulses of the excitation vectors in said first subcodebook, and computation means for obtaining the excitation information using the gain controlled excitation

vectors.

3. The apparatus according to claim 1, wherein said control means makes the gain for the excitation vectors in said second subcodebook relatively small in a case where the distance between pulses of excitation vectors in said first subcodebook is short, while makes the gain for the excitation vectors in said second subcodebook relatively large in another case where the distance between pulses of excitation vectors in said first subcodebook is long.

4. The apparatus according to claim 3, wherein said control means calculates the gain according to a following equation 1,

$$g = |P1 - P2| / L \quad \dots \text{equation (1)}$$

wherein g is the gain, $P1$ and $P2$ are respectively excitation vector positions in first subcodebook, and L is a vector length.

5. The apparatus according to claim 1, said stochastic codebook further has instruction means for instructing an excitation vector to be acquired from said first subcodebook and said second subcodebook corresponding to a distance between excitation vectors in said first subcodebook, and switching means for switching between outputs of the excitation vectors in said first subcodebook and said second subcodebook according to the instruction by said instruction means.

6. An apparatus for performing speech coding in a CELP system, said apparatus comprising:

an adaptive codebook in which previously synthesized

excitation signals are stored;

a stochastic codebook in which a plurality of excitation vectors are stored, said stochastic codebook having a first subcodebook in which excitation vectors composed of a small number of pulses are stored and a second subcodebook in which excitation vectors composed of a large number of pulses are stored;

means for obtaining a synthesized speech using excitation information acquired from said adaptive codebook and said stochastic codebook, using LPC obtained by performing LPC analysis on an input speech signal;

means for executing a voiced/unvoiced judgment on said input speech signal using said LPC;

means for obtaining gain information for said synthesized speech using a relation of said synthesized speech and said input speech signal; and

means for transmitting said LPC, said excitation information and said gain information.

7. The apparatus according to claim 6, wherein said stochastic codebook further has control means for controlling a gain for respective excitation vectors in said first subcodebook and said second subcodebook corresponding to a distance between pulses of the excitation vector in said first subcodebook, and computation means for obtaining the excitation information using the gain controlled excitation vectors.

8. The apparatus according to claim 6, wherein said

control means makes the gain for the excitation vector in said second subcodebook relatively small in a case where the distance between pulses of excitation vectors in said first subcodebook is short, while makes the gain for the excitation vector in said second subcodebook relatively large in another case where the distance between pulses of excitation vectors in said first subcodebook is long.

9. The apparatus according to claim 7, wherein said control means calculates the gain according to a following equation 2,

$$g = |P1 - P2| / R \quad \dots \text{equation (2)}$$

wherein g is the gain, $P1$ and $P2$ are respectively excitation vector positions in said first subcodebook, and R represents a weighting coefficient and is a vector length L in a case where a result of the voiced/unvoiced judgment indicates a voiced speech, and $L \times 0.5$ in another case where the result of the voiced/unvoiced judgment indicates an unvoiced speech.

10. The apparatus according to claim 6, said stochastic codebook further has instruction means for instructing an excitation vector to be acquired from said first subcodebook and said second subcodebook corresponding to a distance between excitation vectors of said first subcodebook, and switching means for switching between outputs of the excitation vectors in said first subcodebook and said second subcodebook according to the instruction by said instruction means.

11. An apparatus for performing speech decoding in a

CSLP system, said apparatus comprising:

an adaptive codebook in which previously synthesized excitation signals are stored;

a stochastic codebook in which a plurality of excitation vectors are stored, said stochastic codebook having a first subcodebook in which excitation vectors composed of a small number of pulses are stored and a second subcodebook in which excitation vectors composed of a large number of pulses are stored;

10 means for receiving LPC, excitation information and gain information transmitted from a coding side; and

means for decoding a speech using said excitation information multiplied by said gain information, and said LPC .

12. The apparatus according to claim 11, wherein said apparatus further comprises means for providing said LPC to said stochastic codebook .

13. A method for performing speech coding in a CELP system, said method comprising the steps of:

controlling a gain for respective excitation vectors in a first subcodebook and a second subcodebook corresponding to a distant between pulses of excitation vectors in said first subcodebook of a stochastic codebook having said first subcodebook in which excitation vectors composed of a small number of pulses are stored and said second subcodebook in which excitation vectors composed of a large number of pulses are stored;

obtaining excitation information using gain controlled

excitation vectors;

obtaining a synthesized speech using excitation information acquired from an adaptive codebook and said stochastic codebook, using LPC obtained by performing LPC
5 analysis on an input speech signal; and

obtaining gain information for said synthesized speech using a relation of said synthesized speech and said input speech signal.

14. The method according to claim 13, wherein said
10 method further comprises the step of performing a voiced/unvoiced judgment on said input speech signal using said LPC.

15. A method for performing speech coding in a CELP system, said method comprising the steps of:

15 selecting an excitation vector in either of a first subcodebook or a second subcodebook corresponding to a distant between pulses of excitation vectors in said first subcodebook of a stochastic codebook having said first subcodebook in which excitation vectors composed of a small number of pulses are
20 stored and said second subcodebook in which excitation vectors composed of a large number of pulses are stored;

obtaining excitation information using the selected excitation vector;

obtaining a synthesized speech using excitation
25 formation acquired from an adaptive codebook and said stochastic codebook, using LPC obtained by performing LPC analysis on an input speech signal; and

obtaining gain information for said synthesized speech using a relation of said synthesized speech and said input speech signal.

16. The method according to claim 15, wherein said
5 method further comprises the step of performing a voiced/unvoiced judgment on said input speech signal using said LPC.

17. A recording medium readable by a computer, said
medium storing a speech coding program, an adaptive codebook
10 in which previously synthesized excitation signals are stored, and a stochastic codebook in which a plurality of excitation vectors are stored, said stochastic codebook having a first subcodebook in which excitation vectors composed of a small number of pulses are stored and a second subcodebook in which
15 excitation vectors composed of a large number of pulses are stored, said program including the procedures of:

controlling a gain for respective excitation vectors in
said first subcodebook and said second subcodebook
corresponding to a distant between pulses of excitation
20 vectors in said first subcodebook of said stochastic codebook;

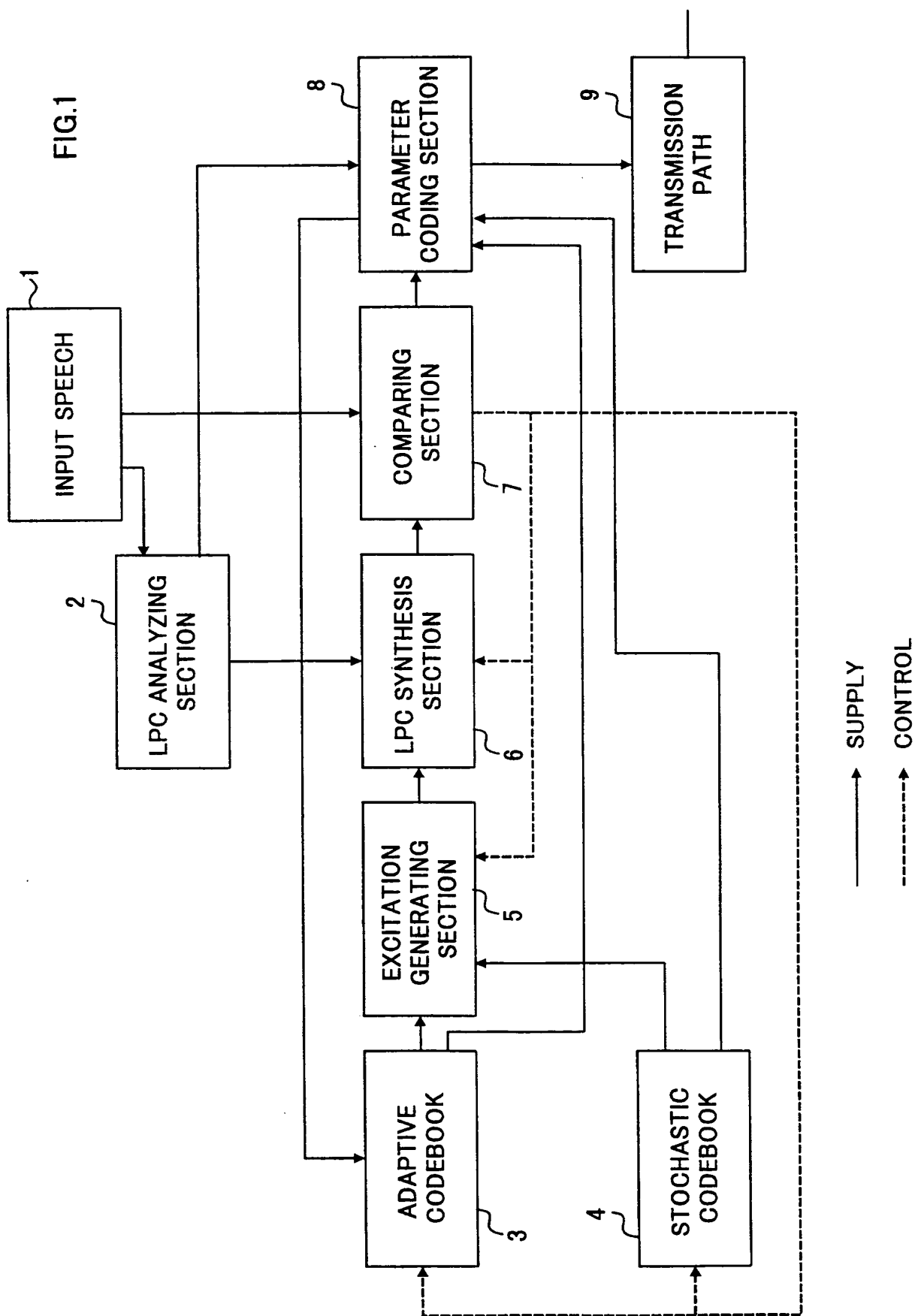
obtaining excitation information using gain controlled excitation vectors;

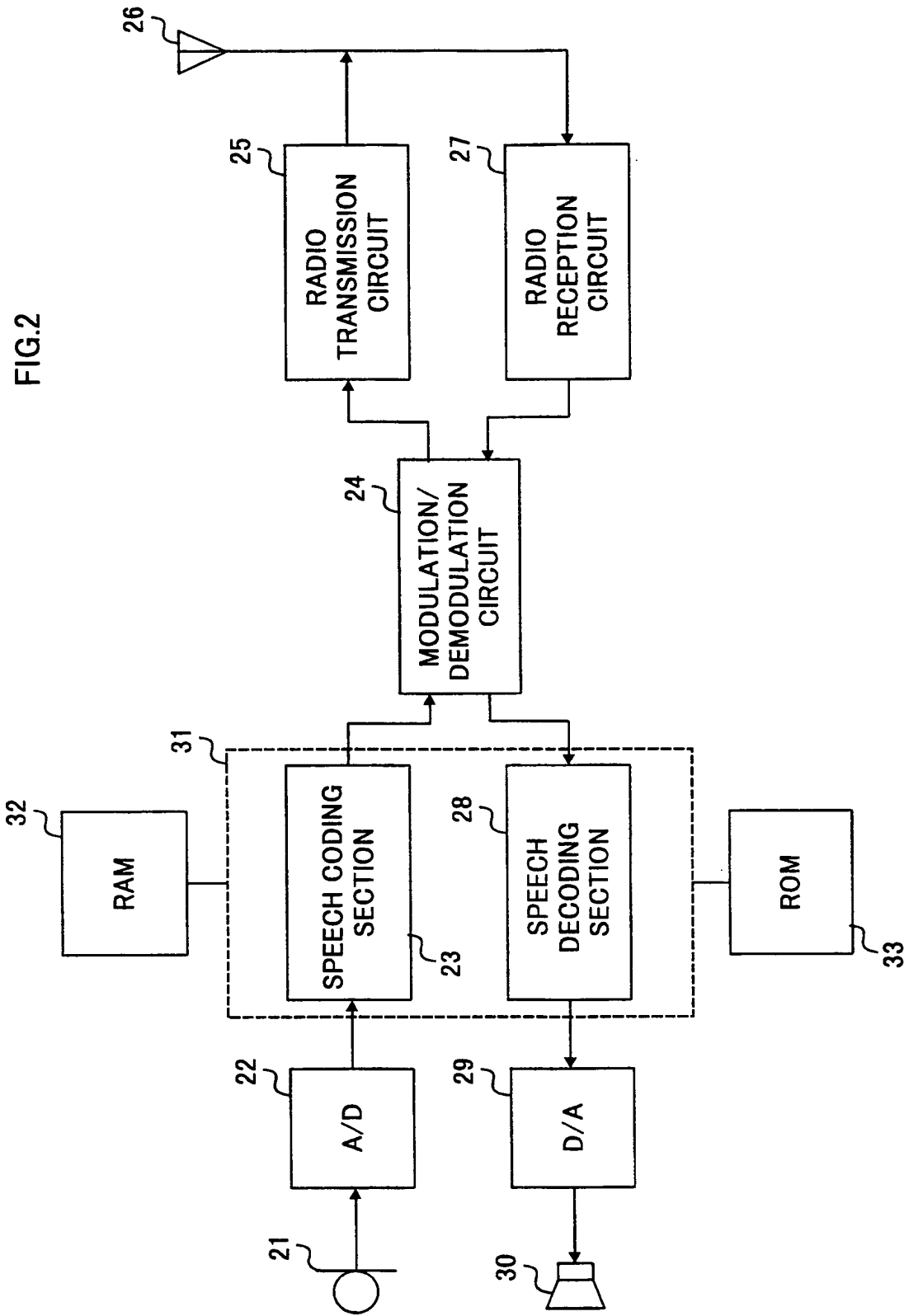
obtaining a synthesized speech using excitation
information acquired from said adaptive codebook and said
25 stochastic codebook, using LPC obtained by performing LPC analysis on an input speech signal; and

obtaining gain information for said synthesized speech

using a relation of said synthesized speech and said input speech signal.

1/12





3 / 12

FIG.3

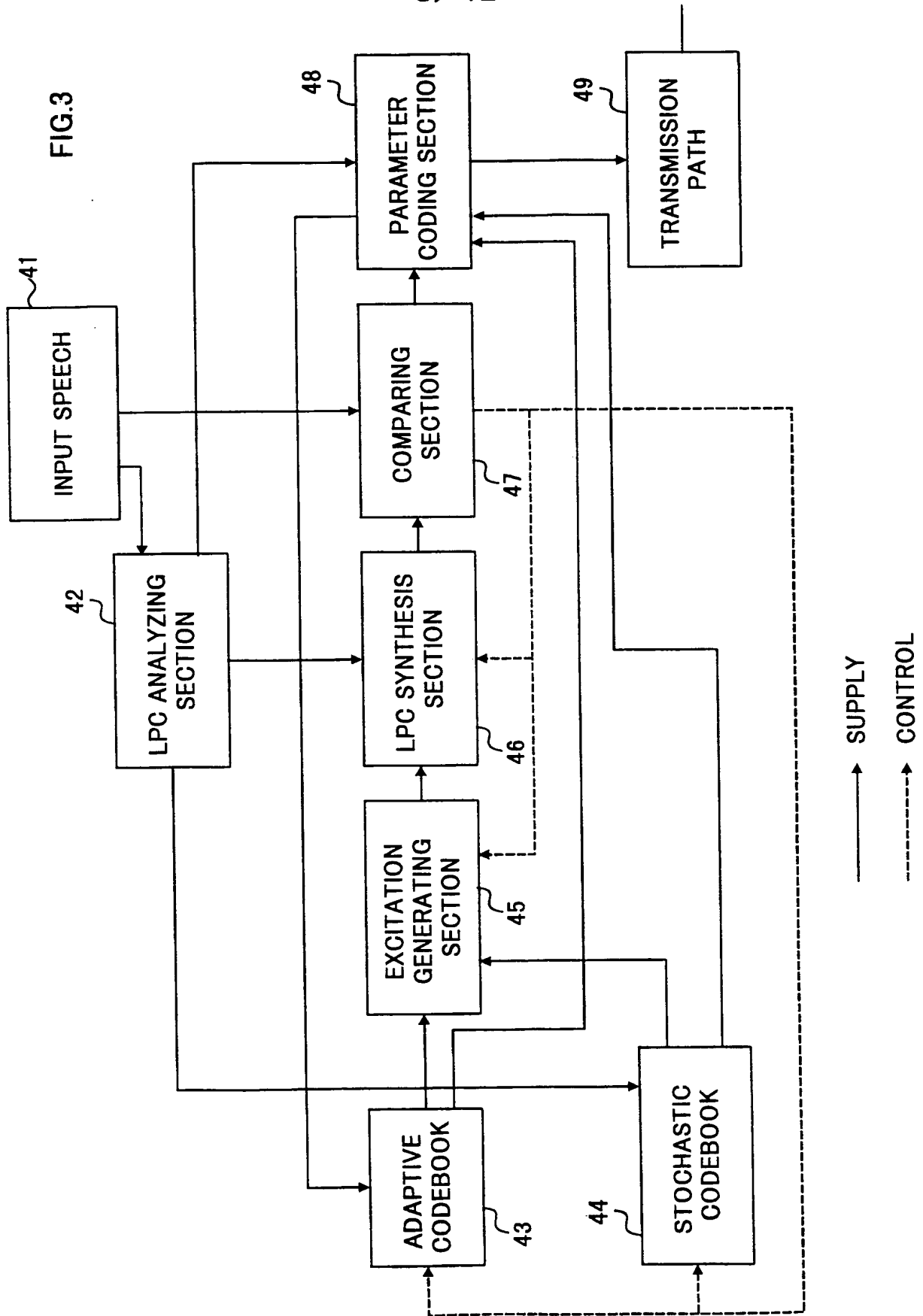


FIG.4

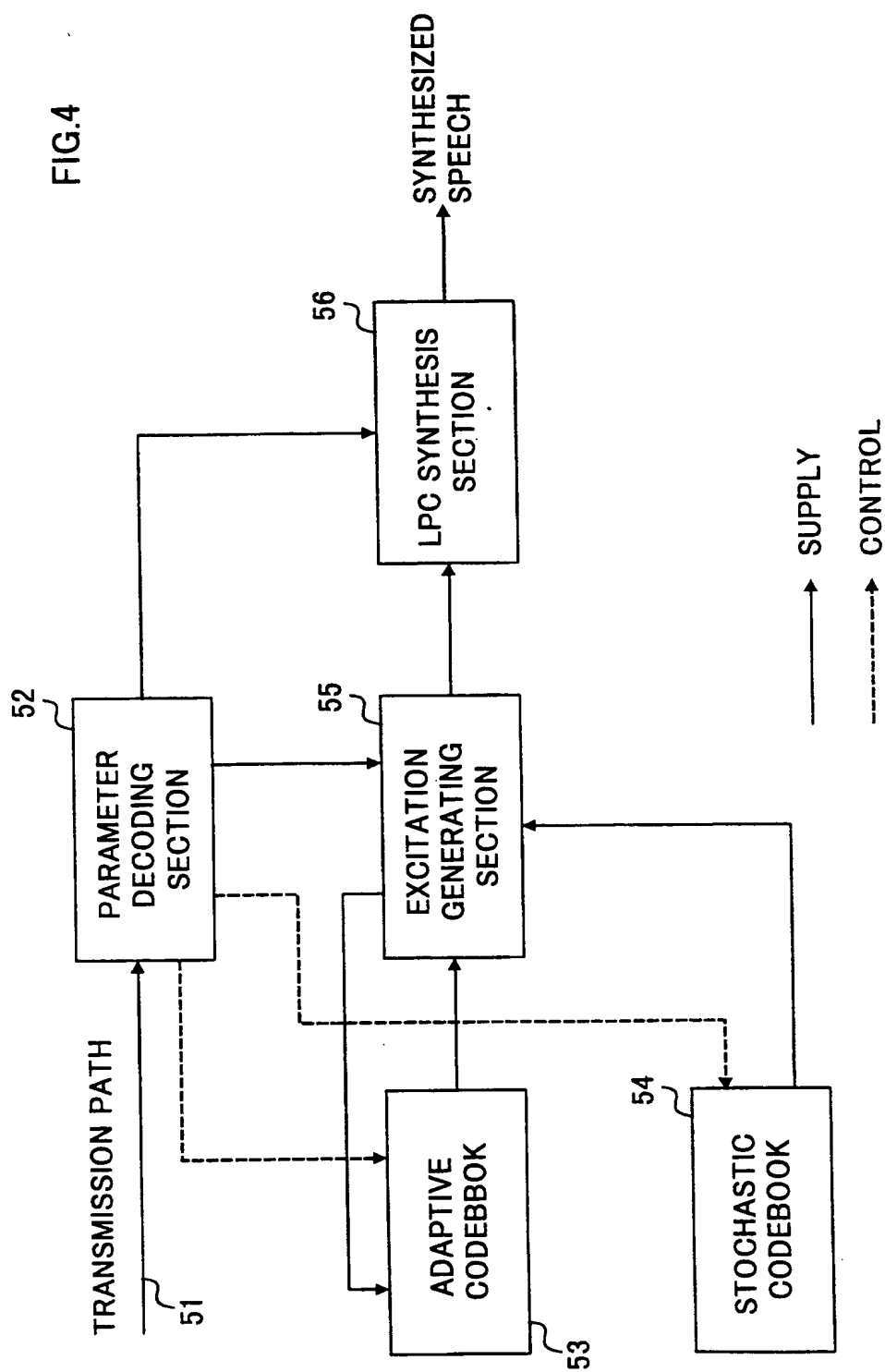


FIG.5

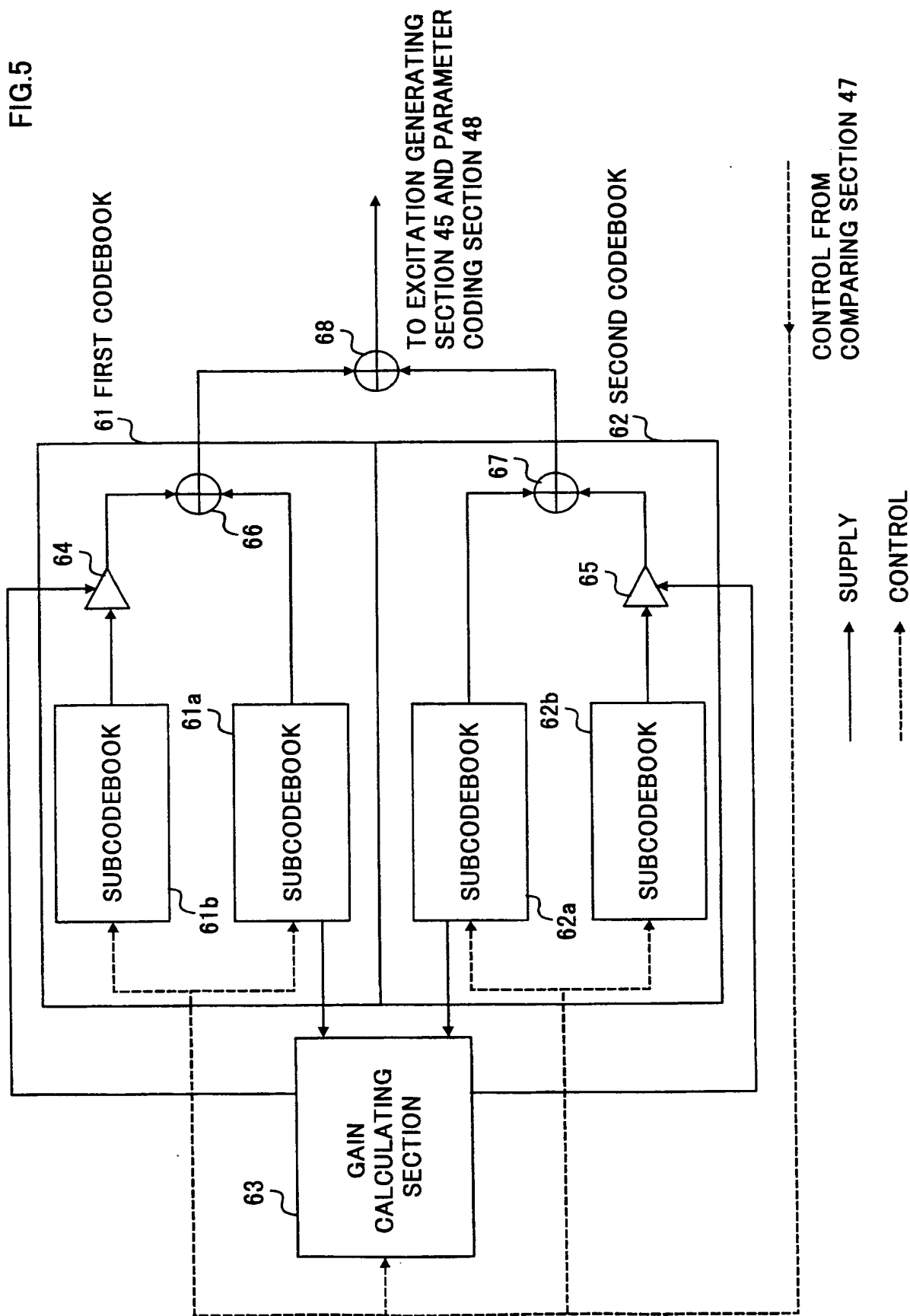


FIG. 6A

61a SUBCODEBOOK

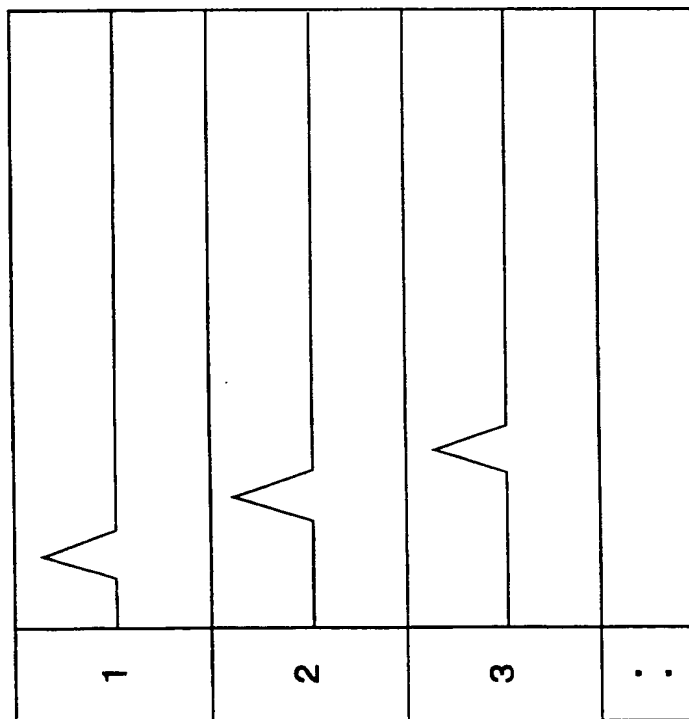
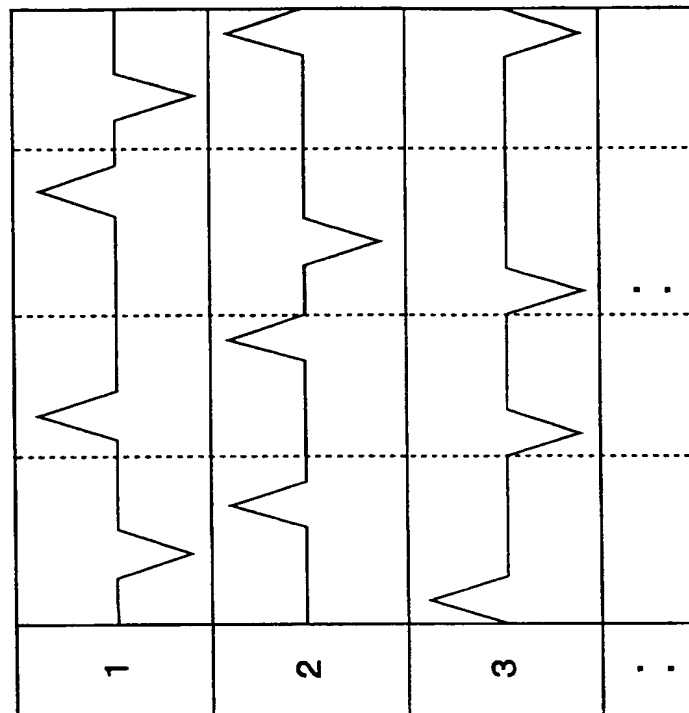
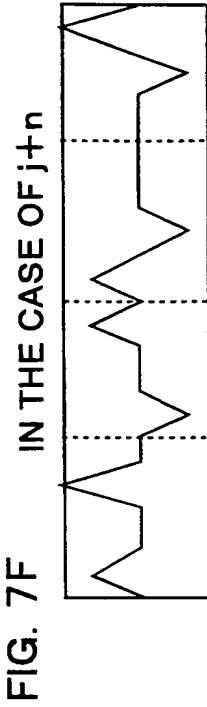
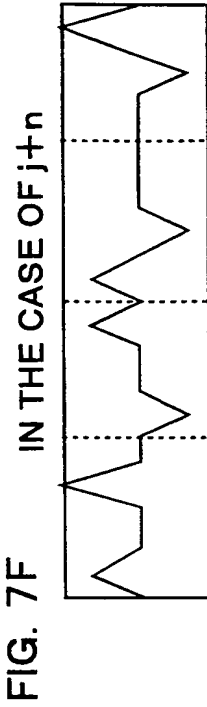
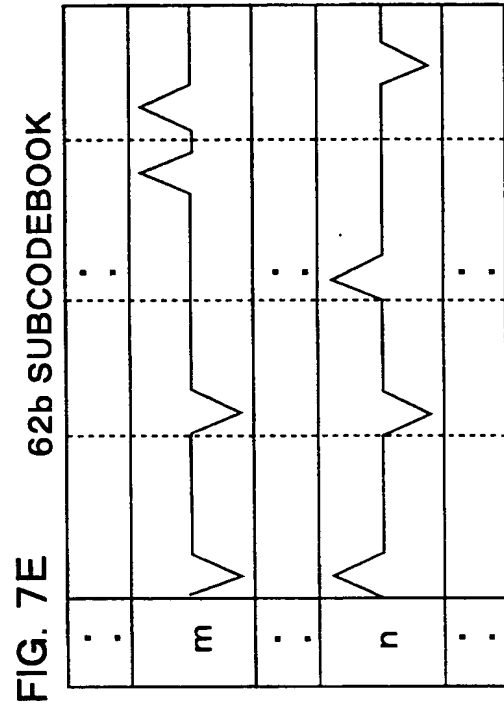
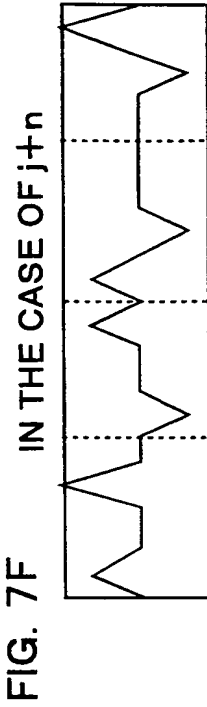
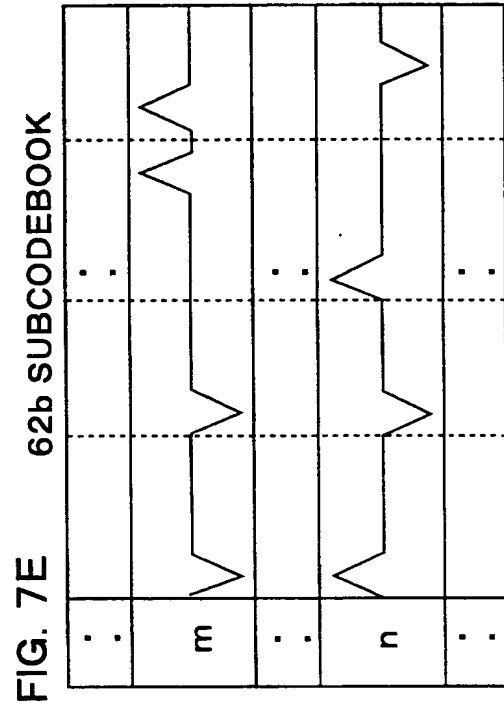
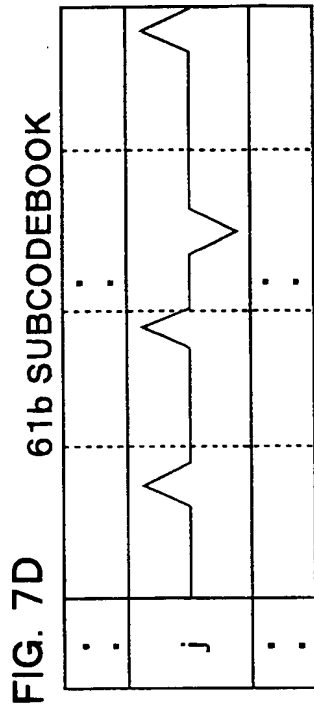


FIG. 6B

61b SUBCODEBOOK





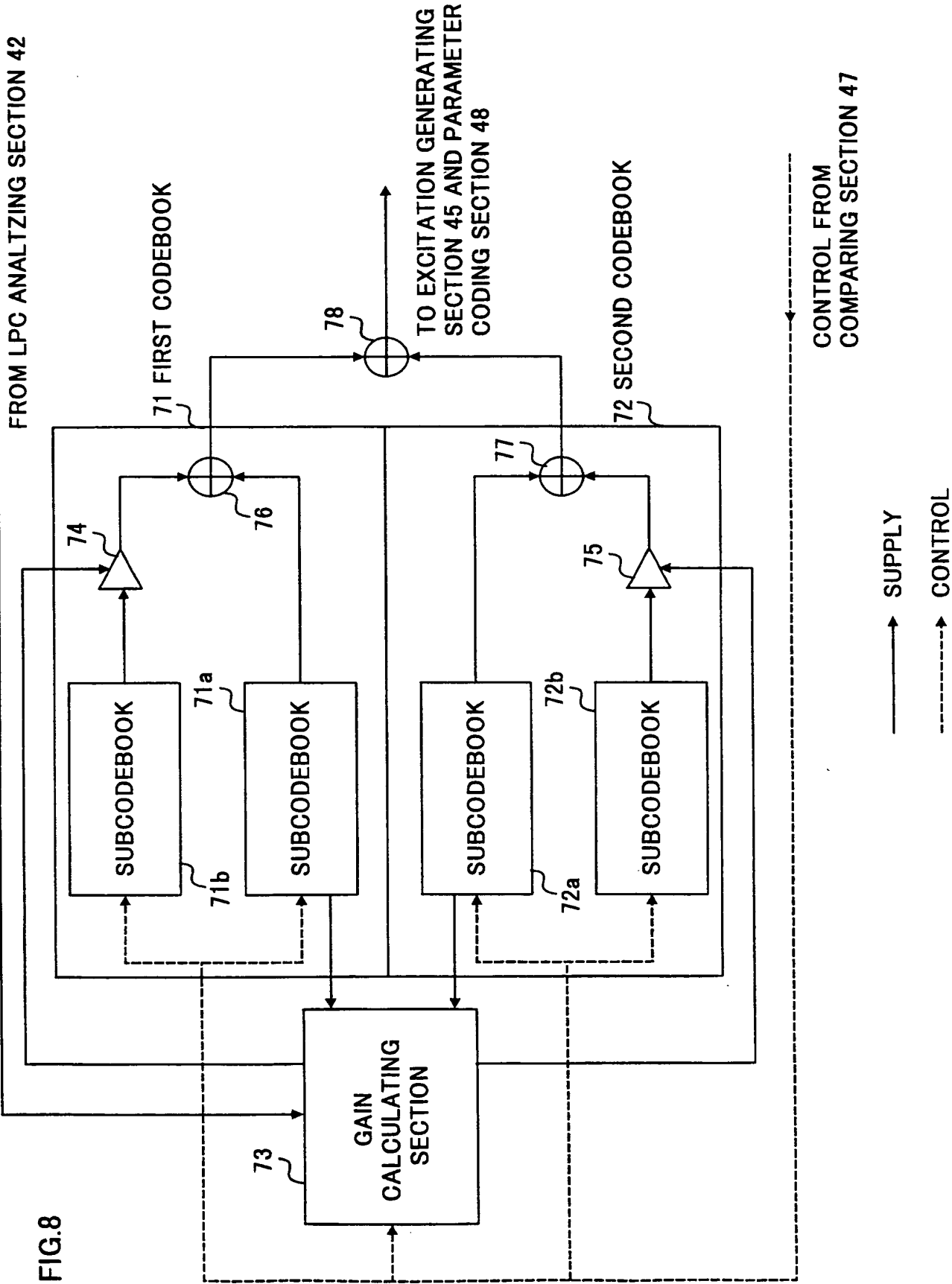


FIG.9

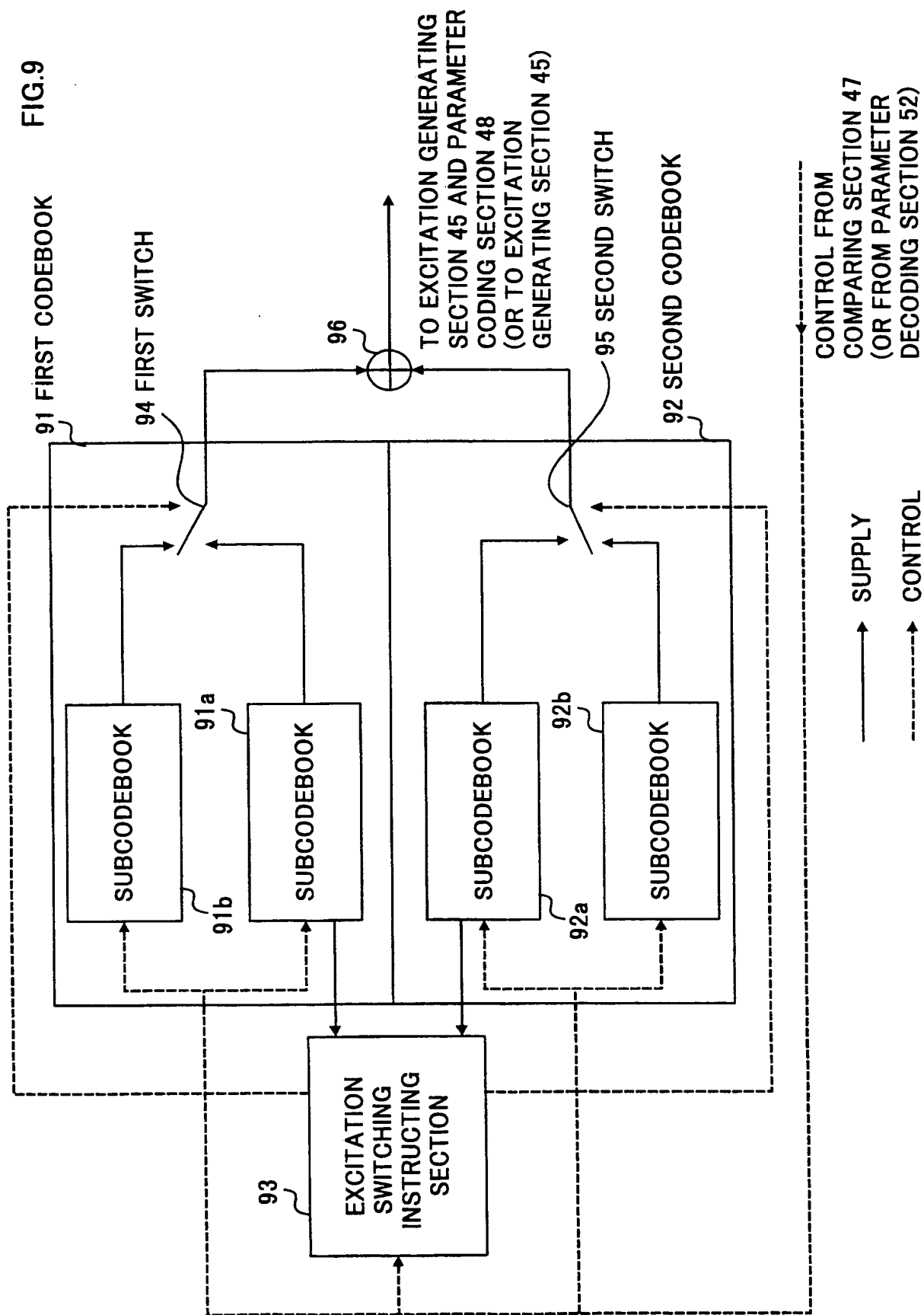


FIG. 10A

91a SUBCODEBOOK

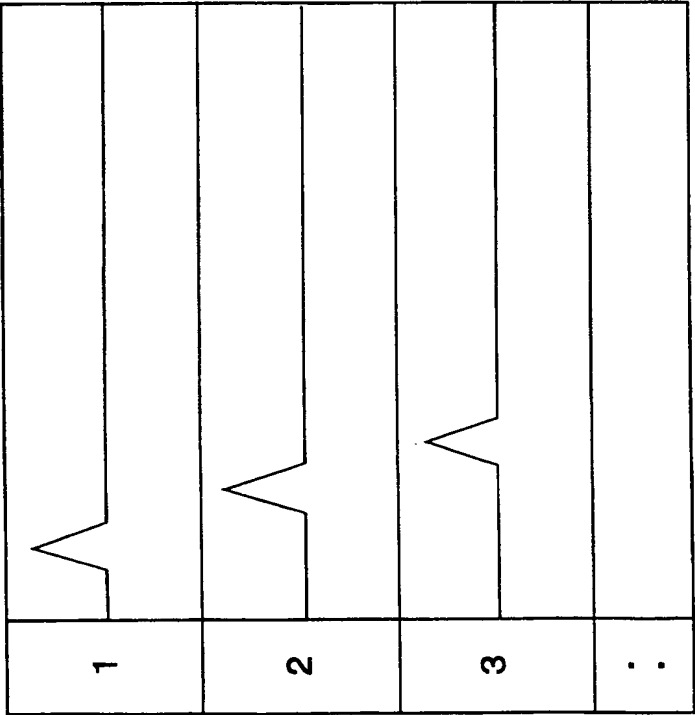


FIG. 10B

91b SUBCODEBOOK

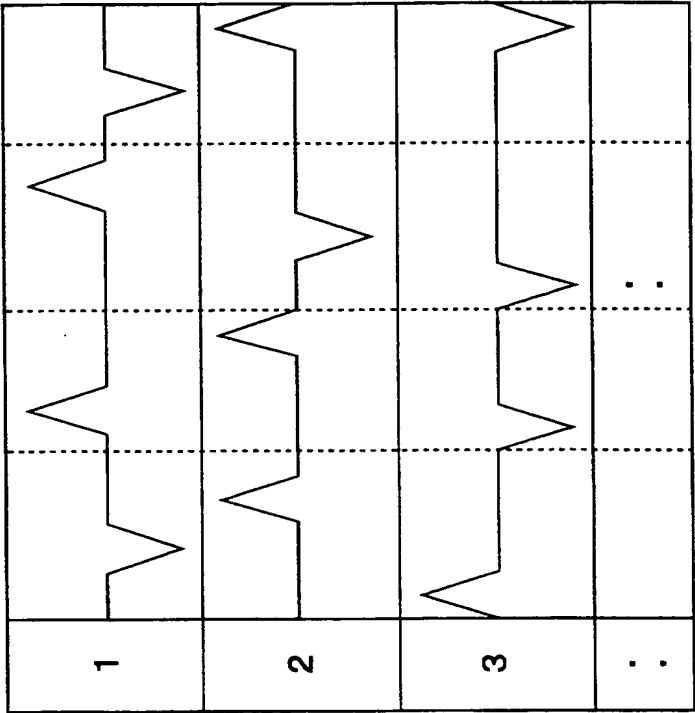


FIG. 11A 91a SUBCODEBOOK

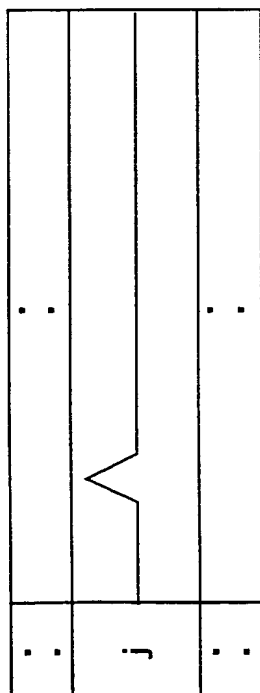


FIG. 11D 91b SUBCODEBOOK

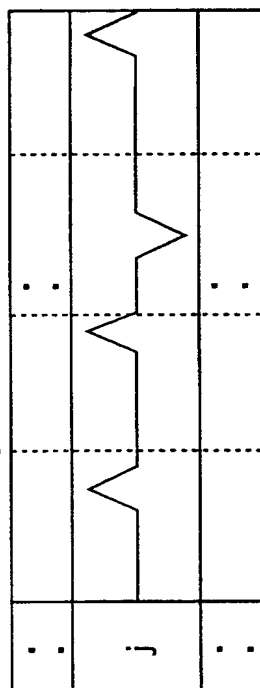


FIG. 11B 92a SUBCODEBOOK

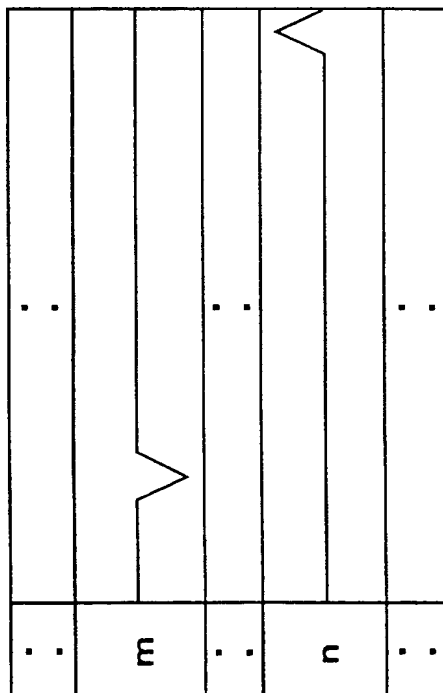


FIG. 11E 92b SUBCODEBOOK

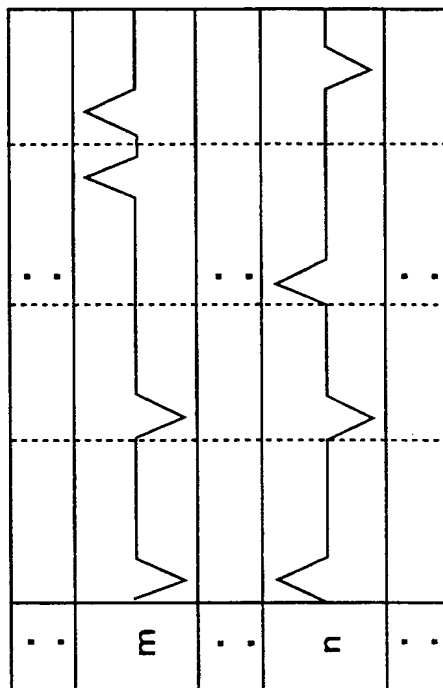


FIG. 11C IN THE CASE OF j+m

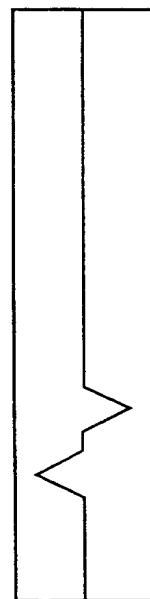


FIG. 11F IN THE CASE OF j+n

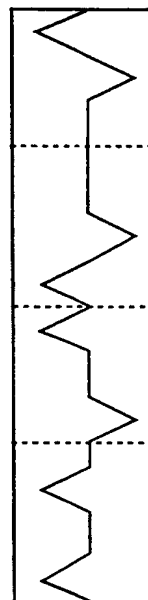
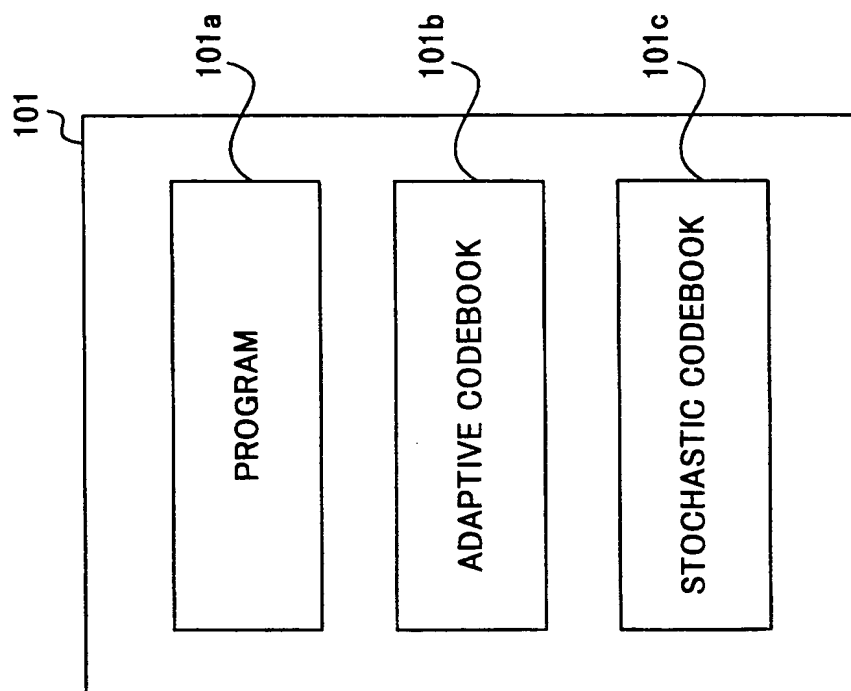


FIG.12



INTERNATIONAL SEARCH REPORT

International Application No

PCT/JP 99/03064

A. CLASSIFICATION OF SUBJECT MATTER

IPC 6 G10L9/14

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 95 16260 A (PACIFIC COMM SCIENCES INC) 15 June 1995 (1995-06-15) abstract; claims 1-7	1,5,13, 15
A	---	17
Y	WO 91 13432 A (UNIV SHERBROOKE) 5 September 1991 (1991-09-05) abstract page 2, line 20 - line 31 page 14, line 1 - line 21 ---	1,5,13, 15
	-/--	

☒ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

Special categories of cited documents:

- "A" document defining the general state of the art which is not considered to be of particular relevance
- "E" earlier document but published on or after the international filing date
- "L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
- "O" document referring to an oral disclosure, use, exhibition or other means
- "P" document published prior to the international filing date but later than the priority date claimed

- "T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
- "X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
- "Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
- "&" document member of the same patent family

Date of the actual completion of the international search

21 September 1999

Date of mailing of the international search report

28/09/1999

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
Fax: (+31-70) 340-3016

Authorized officer

Van Doremalen, J

INTERNATIONAL SEARCH REPORT

International Application No

PCT/JP 99/03064

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	<p>AKAMINE M ET AL: "ADAPTIVE DENSITY PULSE EXCITATION FOR LOW BIT RATE SPEECH CODING" IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS, COMMUNICATIONS AND COMPUTER SCIENCES, vol. E78-A, no. 2, 1 February 1995 (1995-02-01), pages 199-207, XP000515192 ISSN: 0916-8508 abstract page 199, right-hand column, line 43 - line 49</p>	1,6,11, 13,15,17
A	<p>MASAMI AKAMINE ET AL: "CELP CODING WITH AN ADAPTIVE DENSITY PULSE EXCITATION MODEL" SPEECH PROCESSING 1, ALBUQUERQUE, APRIL 3 - 6, 1990, vol. 1, no. CONF. 15, 3 April 1990 (1990-04-03), pages 29-32, XP000146401 INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS abstract paragraph '02.2!</p>	1,6,11, 13,15,17

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No

PCT/JP 99/03064

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 9516260 A	15-06-1995	US 5717824 A	10-02-1998
		AU 1336995 A	27-06-1995
		CA 2178073 A	15-06-1995
		EP 0733257 A	25-09-1996
WO 9113432 A	05-09-1991	CA 2010830 A	23-08-1991
		AT 164252 T	15-04-1998
		AU 6632890 A	18-09-1991
		DE 69032168 D	23-04-1998
		DE 69032168 T	08-10-1998
		EP 0516621 A	09-12-1992
		ES 2116270 T	16-07-1998
		US 5699482 A	16-12-1997
		US 5754976 A	19-05-1998
		US 5701392 A	23-12-1997
		US 5444816 A	22-08-1995

RECEIVED
AUG 30, 1999
Washida & Associates

PATENT COOPERATION TREATY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION CONCERNING
SUBMISSION OR TRANSMITTAL
OF PRIORITY DOCUMENT

(PCT Administrative Instructions, Section 411)

To:

WASHIDA, Kimihito
Shintoshicenter Building
5th floor
24-1, Tsurumaki 1-chome
Tama-shi
Tokyo 206-0034
JAPON

Date of mailing (day/month/year) 17 August 1999 (17.08.99)	
Applicant's or agent's file reference 1F99035-PCT	IMPORTANT NOTIFICATION
International application No. PCT/JP99/03064	
International filing date (day/month/year) 08 June 1999 (08.06.99)	International publication date (day/month/year) Not yet published
Priority date (day/month/year) 09 June 1998 (09.06.98)	Applicant MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD. et al

1. The applicant is hereby notified of the date of receipt (except where the letters "NR" appear in the right-hand column) by the International Bureau of the priority document(s) relating to the earlier application(s) indicated below. Unless otherwise indicated by an asterisk appearing next to a date of receipt, or by the letters "NR", in the right-hand column, the priority document concerned was submitted or transmitted to the International Bureau in compliance with Rule 17.1(a) or (b).
2. This updates and replaces any previously issued notification concerning submission or transmittal of priority documents.
3. An asterisk(*) appearing next to a date of receipt, in the right-hand column, denotes a priority document submitted or transmitted to the International Bureau but not in compliance with Rule 17.1(a) or (b). In such a case, the attention of the applicant is directed to Rule 17.1(c) which provides that no designated Office may disregard the priority claim concerned before giving the applicant an opportunity, upon entry into the national phase, to furnish the priority document within a time limit which is reasonable under the circumstances.
4. The letters "NR" appearing in the right-hand column denote a priority document which was not received by the International Bureau or which the applicant did not request the receiving Office to prepare and transmit to the International Bureau, as provided by Rule 17.1(a) or (b), respectively. In such a case, the attention of the applicant is directed to Rule 17.1(c) which provides that no designated Office may disregard the priority claim concerned before giving the applicant an opportunity, upon entry into the national phase, to furnish the priority document within a time limit which is reasonable under the circumstances.

<u>Priority date</u>	<u>Priority application No.</u>	<u>Country or regional Office or PCT receiving Office</u>	<u>Date of receipt of priority document</u>
09 June 1998 (09.06.98)	10/160119	JP	27 July 1999 (27.07.99)
11 Sept 1998 (11.09.98)	10/258271	JP	27 July 1999 (27.07.99)

The International Bureau of WIPO 34, chemin des Colombettes 1211 Geneva 20, Switzerland Facsimile No. (41-22) 740.14.35	Authorized officer Juan Cruz Telephone No. (41-22) 338.83.38
--	--



INTERNATIONAL SEARCH REPORT

International Application No

PCT/JP 99/03064

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 G10L9/14

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC 6 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 95 16260 A (PACIFIC COMM SCIENCES INC) 15 June 1995 (1995-06-15) abstract; claims 1-7	1,5,13, 15
A	---	17
Y	WO 91 13432 A (UNIV SHERBROOKE) 5 September 1991 (1991-09-05) abstract page 2, line 20 - line 31 page 14, line 1 - line 21 ---	1,5,13, 15
	-/--	

☒ Further documents are listed in the continuation of box C.

☒ Patent family members are listed in annex.

*** Special categories of cited documents:**

- "A" document defining the general state of the art which is not considered to be of particular relevance
- "E" earlier document but published on or after the international filing date
- "L" document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)
- "O" document referring to an oral disclosure, use, exhibition or other means
- "P" document published prior to the international filing date but later than the priority date claimed

- "T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
- "X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
- "Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
- "&" document member of the same patent family

Date of the actual completion of the international search

21 September 1999

Date of mailing of the international search report

28/09/1999

Name and mailing address of the ISA

European Patent Office, P.B. 5818 Patentlaan 2
NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl.
Fax: (+31-70) 340-3016

Authorized officer

Van Doremalen, J

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	AKAMINE M ET AL: "ADAPTIVE DENSITY PULSE EXCITATION FOR LOW BIT RATE SPEECH CODING" IEICE TRANSACTIONS ON FUNDAMENTALS OF ELECTRONICS, COMMUNICATIONS AND COMPUTER SCIENCES, vol. E78-A, no. 2, 1 February 1995 (1995-02-01), pages 199-207, XP000515192 ISSN: 0916-8508 abstract page 199, right-hand column, line 43 - line 49 ---	1,6,11, 13,15,17
A	MASAMI AKAMINE ET AL: "CELP CODING WITH AN ADAPTIVE DENSITY, PULSE EXCITATION MODEL" SPEECH PROCESSING 1, ALBUQUERQUE, APRIL 3 - 6, 1990, vol. 1, no. CONF. 15, 3 April 1990 (1990-04-03), pages 29-32, XP000146401 INSTITUTE OF ELECTRICAL AND ELECTRONICS ENGINEERS abstract paragraph '02.2! -----	1,6,11, 13,15,17



1/4

PCT REQUEST

Original (for SUBMISSION) - printed on 07.06.1999 01:18:26 PM

1F99035-PCT

0	For receiving Office use only	
0-1	International Application No.	
0-2	International Filing Date	
0-3	Name of receiving Office and "PCT International Application"	

0-4	Form - PCT/RO/101 PCT Request Prepared using	PCT-EASY Version 2.83 (updated 01.03.1999)
0-5	Petition The undersigned requests that the present international application be processed according to the Patent Cooperation Treaty	
0-6	Receiving Office (specified by the applicant)	Japanese Patent Office (RO/JP)
0-7	Applicant's or agent's file reference	1F99035-PCT
I	Title of Invention	SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS
II	Applicant	
II-1	This person is:	applicant only
II-2	Applicant for	all designated States except US
II-4	Name	MATSUSHITA ELECTRIC INDUSTRIAL CO., LTD.
II-5	Address:	1006, Oaza Kadoma, Kadoma-shi, Osaka 571-8501 Japan
II-6	State of nationality	JP
II-7	State of residence	JP
II-8	Telephone No.	06-6908-1473
II-9	Facsimile No.	06-6909-0053
III-1	Applicant and/or inventor	
III-1-1	This person is:	applicant and inventor
III-1-2	Applicant for	US only
III-1-4	Name (LAST, First)	MORII, Toshiyuki
III-1-5	Address:	2-3-7-501, Nijigaoka, Asao-ku, Kawasaki-shi, Kanagawa 215-0015 Japan
III-1-6	State of nationality	JP
III-1-7	State of residence	JP

PCT REQUEST

Original (for SUBMISSION) - printed on 07.06.1999 01:18:26 PM

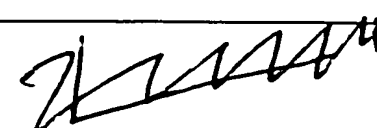
1F99035-PCT

III-2	Applicant and/or inventor	
III-2-1	This person is:	applicant and inventor
III-2-2	Applicant for	US only
III-2-4	Name (LAST, First)	YASUNAGA, Kazutoshi
III-2-5	Address:	3-33-17-305, Sugao, Miyamae-ku, Kawasaki-shi, Kanagawa 216-0015 Japan
III-2-6	State of nationality	JP
III-2-7	State of residence	JP
IV-1	Agent or common representative; or address for correspondence	
	The person identified below is hereby/has been appointed to act on behalf of the applicant(s) before the competent International Authorities as:	agent
IV-1-1	Name (LAST, First)	WASHIDA, Kimihito
IV-1-2	Address:	5th Floor, Shintoshicenter Bldg., 24-1, Tsurumaki 1-chome Tama-shi, Tokyo 206-0034 Japan
IV-1-3	Telephone No.	042-338-4600
IV-1-4	Facsimile No.	042-338-4605
V	Designation of States	
V-1	Regional Patent (other kinds of protection or treatment, if any, are specified between parentheses after the designation(s) concerned)	EP: AT BE CH&LI CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE and any other State which is a Contracting State of the European Patent Convention and of the PCT
V-2	National Patent (other kinds of protection or treatment, if any, are specified between parentheses after the designation(s) concerned)	CA CN JP KR US
V-5	Precautionary Designation Statement In addition to the designations made under items V-1, V-2 and V-3, the applicant also makes under Rule 4.9(b) all designations which would be permitted under the PCT except any designation(s) of the State(s) indicated under item V-6 below. The applicant declares that those additional designations are subject to confirmation and that any designation which is not confirmed before the expiration of 15 months from the priority date is to be regarded as withdrawn by the applicant at the expiration of that time limit.	
V-6	Exclusion(s) from precautionary designations	NONE
VI-1	Priority claim of earlier national application	
VI-1-1	Filing date	09 June 1998 (09.06.1998)
VI-1-2	Number	10-160119
VI-1-3	Country	JP

PCT REQUEST

Original (for SUBMISSION) - printed on 07.06.1999 01:18:26 PM

1F99035-PCT

VI-2	Priority claim of earlier national application		
VI-2-1	Filing date	11 September 1998 (11.09.1998)	
VI-2-2	Number	10-258271	
VI-2-3	Country	JP	
VI-3	Priority document request The receiving Office is requested to prepare and transmit to the International Bureau a certified copy of the earlier application(s) identified above as item(s):	VI-1, VI-2	
VII-1	International Searching Authority Chosen	European Patent Office (EPO) (ISA/EP)	
VIII	Check list	number of sheets	electronic file(s) attached
VIII-1	Request	4	-
VIII-2	Description	39	-
VIII-3	Claims	8	-
VIII-4	Abstract	1	1f99035-pct.txt
VIII-5	Drawings	12	-
VIII-7	TOTAL	64	
	Accompanying items	paper document(s) attached	electronic file(s) attached
VIII-8	Fee calculation sheet	✓	-
VIII-9	Separate signed power of attorney	✓	-
VIII-10	Copy of general power of attorney	✓	-
VIII-16	PCT-EASY diskette	-	diskette
VIII-17	Other (specified):	Translation of copy of general power of attorney	-
VIII-17	Other (specified):	Request for Transmittal of Priority Document	-
VIII-17	Other (specified):	Evidence of Bank Draft	-
VIII-17	Other (specified):	Sheet being accompanied by Revenue Stamps to be paid	-
VIII-18	Figure of the drawings which should accompany the abstract	5	
VIII-19	Language of filing of the International application	English	
IX-1	Signature of applicant or agent		
IX-1-1	Name (LAST, First)	WASHIDA, Kimihito	

FOR RECEIVING OFFICE USE ONLY

10-1	Date of actual receipt of the purported International application	
10-2	Drawings:	
10-2-1	Received	
10-2-2	Not received	

PCT REQUEST

Original (for SUBMISSION) - printed on 07.06.1999 01:18:26 PM

1F99035-PCT

10-3	Corrected date of actual receipt due to later but timely received papers or drawings completing the purported international application	
10-4	Date of timely receipt of the required corrections under PCT Article 11(2)	
10-5	International Searching Authority	ISA/EP
10-6	Transmittal of search copy delayed until search fee is paid	

FOR INTERNATIONAL BUREAU USE ONLY

11-1	Date of receipt of the record copy by the International Bureau	
------	--	--